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Translation

PROCEEDINGS OF THE FOURTH ALL-UNION SYMPOSIUM ON CONTROL PROBLEMS IN COMMUNICATIONS NETWORKS AND CENTERS

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Moscow TRUDY IV VSESOYUZNOGO SIMPOZIUMA PO PROBLEMAM UPRAVLENIYA NA SETYAKH I UZLAKH SVYAZI in Russian 1981 (signed to press 26 Feb 81) pp 1-156

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CONTENTS

ANNOTATION

[Text] The given collection includes expanded summaries of the reports presented at the Fourth All-Union Symposium on the control problems of communications networks, the construction of dynamic information flow control systems, the problems of introducing control systems in the communications networks and centers of the USSR and the construction of programmed control devices. In addition, a study is made of the analytical methods of analyzing networks with dynamic traffic control and the problems of their application in practice.

PROBLEM OF DETERMINING THE LOAD ON CENTRALIZED CONTROL UNITS AT SWITCHING CENTERS

[Article by R. A. Avakov, V. D. Safronov, Leningrad, pp 3-5]

When developing centralized control units (CCU) based on control computers (EUM), a number of peculiarities arise in determining the incoming and serviced load. These peculiarities are caused by the multiple staging, the presence of priorities and temporary restrictions in servicing the calls [1-3].

The incoming telephone call traffic creates a load on the switching center (SC). The load serviced by the switching center with given subscriber servicing quality norms determines the carrying capacity of the switching center as a whole. The carrying capacity of a switching center with programmed control depends on the carrying capacity of the switching system (SS) and the centralized control units (CCU). Let us stipulate that the carrying capacity of the CCU be expressed in terms of its output capacity, which will be determined by the maximum number of serviced calls per unit time under the condition of the presence of a sufficient number of SS units. Thus, the output capacity of the CCU can be expressed in terms of the load for which it has the meaning of efficient application of the CCU.

The servicing of calls in the switching center with program control is divided into a number of stages which are determined by the type of communications, the type of communication area, the actions of the calling and called subscribers. Each stage of servicing the calls corresponds to setting up a call (disconnecting) in the switching system between the incoming and outgoing equipment (lines). The flow of calls coming to the switching system "multiplies" on the basis of multistaging and limitation of the functions of the SS sets, and it determines the flow of requests to the CCU. The requests to the CCU come from sources which are the various types of SS sets. The process of servicing the requests consists in successive processing of calls in the CCU-SS section [4]. There is a dependence of the load (Y_{CCU}) coming to the CCU on the load (Y_{SS}) coming to the SS: $Y_{CCU}=f(Y_{SS})$. As Y_{SS} increases, Y_{CCU} also increases; CCU is considered to have exhausted its output capacity when it cannot service an additional load without having a negative effect on the subscriber service quality. Here the difference between the incoming load and the load serviced by the CCU will determine the service losses of the flow of requests ΔY [1]. In reference [5], let us present the possible methods of estimating the carrying capacity of the SS with programmed control.

From the point of view of program processing, each stage of servicing is a separate problem which is realized in the CCU by means of three processes: information reception, processing and output. Representing the operation of the CCU in the form of a single-routing queueing system for each of the stages, it is possible to obtain all the necessary characteristics (waiting time, busy period, and so on) of each of the processes which are required to estimate the load on the CCU.

A formalized description of the process of servicing the CCU calls can be represented in the form of a graph:

$$G = \{A, B\},$$

where $A=\{a_1, \ldots, a_n\}$ is the set of apexes of the graph which correspond to the call servicing stages; $B=\{b_{ij}\}$ is the set of sides of the graph which reflect the information interrelations of the call servicing stages and are given by the matrix:

$$B = \|b_{ij}\| \qquad b_{ij} = \begin{cases} 1 & \text{for } i=j, \\ b_{ij} & \text{in the presence of a side,} \\ 0 & \text{in the absence of a side.} \end{cases}$$

From the matrix B, which is a mapping of the graph G, the set of routes traveled by the call servicing stages between apexes a_i and a_j, $i, j=1,\ldots,n$ is determined. The call servicing route (sequence) is also given by the transmission matrix:

$$P = \|p_{ij}\|,$$

where p_{ij} is the probability that after servicing at the apex a_i the call will go to apex a_j for servicing.

Each apex of the graph G forms its own flow of requests to the CCU in accordance with the load intensity of the i-th flow (λ_i). The flow of requests coming to the CCU for servicing is characterized by the times of their arrival t_i and the servicing time of the requests in different stages $\Delta\tau_i$. The times t_i are determined by the actions of the subscribers (picking up the receiver, beginning to dial the number, and so on). For determining $\Delta\tau_i$ through a given SS it is necessary to know the number of operations executed in the i-th stage k_i (it is determined by the logical diagram of the algorithm) and the duration of each operation Δt_i (it corresponds to the time spent on performing an elementary operation):

$$\Delta\tau_i = k_i \Delta t_i.$$

The flow of requests reaching the CCU creates the load:

$$Y_{CCU} = \sum_{i=1}^{n} \lambda_i k_i \Delta t_i.$$

In addition to the basic time $t_{proc}=\sum_{i=1}^{n} \Delta\tau_i$ spent on processing the requests to the CCU, it is possible to isolate constant time expenditures on the performance of systems operations t_{so}, the operations of scanning the SS sets t_{scan} and the variable time expenditures connected with the maintenance servicing of the system, t_{main}. All of these components can be related by the coefficients q defining the proportion of them out of the total operating time T:

$$t_{SO}=q_1T;\ t_{scan}=q_2T;\ t_{proc}+t_{main}=q_3T.$$

Considering the coefficient α, operation of the CCU without overloads is guaranteed:

$$q_1+q_2+q_3=\alpha T.$$

The time expenditures t_{so} and t_{scan} create a constant load on the CCU independent of the number of incoming requests. When determining the load connected with processing the requests, it is necessary to consider the constant component which takes up a significant proportion of the total load on the CCU.

A study is made of the estimation of the effectiveness of functional organization of request servicing processes in the CCU by the load criterion. The optimization of the call servicing processes is carried out by minimizing the nonproductive machine time expenditures. Determination of the load permits estimation of the duration of the CCU and generation of recommendations for effective use of the EUM in the switching centers.

For proper calculation of the switching center control equipment it is necessary to have systematic statistical data. The formation of the statistical information on the call service quality in the CCU can be carried out on the basis of the load calculation programs.

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ESTIMATING THE OPERATING EFFICIENCY OF THE CONTROL COMPLEX OF A SWITCHING CENTER

[Article by E. Kh. Allayev, M. Sh. Zakhitsov and A. S. Saibov, Tashkent, pp 5-9]

The control system of the switching center must operate continuously over a prolonged period, for failure of the control system can lead to interruption of the operation of the entire system. In order to insure high reliability of the control system, the two-computer principle of switching center control is used [1].

The two-computer control system (DUK) is made up of two identical control computers (EUM). Depending on the selected operating conditions, either one EUM or both of the EUM simultaneously can be used to service the incoming call traffic.

The operation of the DUK can be organized in the following modes:

Asynchronous (s=1),

Synchronous (s=2),

Load separation (s=3),

Source separation (s=4).

In the asynchronous mode of operation of the DUK, the functions connected with servicing the incoming calls are performed by the active EUM (EUM-A), the other (EUM-B) is in "hot" reserve [2]. In the case of failure of an active EUM, the reserve EUM takes on the functions of controlling the switching center. The operating fitness of EUM-A and EUM-B is monitored autonomously using hardware and software (logical program or test) monitoring means. In contrast to the hardware, the software monitoring is accomplished periodically with a period of T_{per}, and it requires expenditures of EUM time equal to τ_{per}. After detection of a failure in the active EUM, the reserve is put into operation after a time interval t_{res}.

In the synchronous mode of operation of the DUK, each incoming call is processed in parallel by two EUM [3]. The results of operation of the EUM-A and EUM-B are compared in order to monitor the fitness of the EUM in defined phases of call processing or periodically (with a period of T_{comp}). In the case of noncomparison of the results of the EUM-A and the EUM-B, each of them includes the monitoring programs providing for finding the failed EUM. Before repair of the failed EUM, the fitness of the EUM that is in working order is checked by the hardware and software.

In the load division mode the incoming call traffic is also serviced in parallel by the EUM-A and the EUM-B [4]. However, in contrast to the synchronous mode, the EUM-A and the EUM-B process different calls. On failure of one of the EUM, the entire load is picked up by the other. After elimination of the cause of failure, the initial structure of the DUK is restored. In the mode with load separation, the fitness of the EUM-A and EUM-B is also checked autonomously using hardware and software monitoring means.

The mode with division of sources is characterized by the fact that all of the load sources of the switching center are divided into two groups. The calls coming from the first group of sources are serviced using the EUM-A, and the calls from the second group of sources, by the EUM-B. On failure of one EUM, the other (the one in good repair) services calls coming from all the load sources. The fitness of the EUM-A and the EUM-B is checked autonomously using hardware and software.

The above-investigated operating conditions of the DUK are not equivalent, for they provide for different output capacity and require different expenditures for organization of the interaction of the EUM. Therefore the problem of determining the operating mode of the DUK in which maximum output capacity will be insured with low expenditures on hardware and software for organization of the interaction of the EUM is of great interest. For selection of this operating mode of the DUK, the quantitative characteristic is required which permits comparison of different versions of the DUK among each other. The output capacity, that is, the number of calls serviced by the DUK with observation of quality indices per unit time, can be used as this characteristic. Since the calls are serviced by the DUK with waiting, and the call waiting time characterizes the operating quality of the system, the output capacity of the DUK can be defined as

$$\Pi = [1 - P\{t > t_a\}] \cdot \Lambda, \quad (1)$$

where Λ is the intensity of the incoming call traffic; $P\{t>t_a\}$ is the probability that the call waiting time will exceed the value of t_a; t_a is the admissible call waiting time.

For selection of the optimal operating mode of the DUK using the index (1), it is first necessary to define the call waiting time distribution functions $W(t)$. The function $W(t)$ depends on many factors: the arrival intensity Λ and the servicing intensity μ of the calls, the failure intensity Λ_0 and repair intensity μ_{rep} of the EUM, the period T_{per} and the time for performance of the check tests τ_{per}, the period T_{comp} and the comparison time τ_{comp} of the results of the EUM-A and EUM-B, and the operating mode of the DUK, s.

An efficient operating mode for the DUK can be selected simply without considering the nature of the incoming traffic or the time required to service the incoming calls. In this case, it is expedient to use the use coefficient of the computing capability of the DUK $\eta^{(s)}$ as the optimalness index of the operating mode of the

DUK. This index is defined as the ratio of the total useful operating time of the EUM-A and the EUM-B $T^{(s)}$ to the total operating time of the two computers 2T in the operating time interval of the DUK T, that is

$$\eta^{(s)}=\frac{T^{(s)}}{2T}. \qquad (2)$$

The operating time of the DUK T consists of three terms:

$$T=t_1+t_2+t_3, \qquad (3)$$

where t_1 is the time during which both EUM are in proper working order; t_2 is the time during which only one EUM is in proper working order; t_3 is the time during which both EUM have failed.

Let us propose that the probabilities of the state of the DUK P_{ij} (i=0.1 and j=0.1) are known, where i characterizes the state of the EUM-A, and j characterizes the state of the EUM-B, where 0 is the working state of the corresponding EUM, and 1 is the failed state; then

$$t_1=T\cdot P_{00},\ t_2=T(P_{01}+P_{10}),\ t_3=T\cdot P_{11}. \qquad (4)$$

Let us consider the time during which the EUM can be busy with servicing incoming calls as the useful operating time of the DUK $T^{(s)}$. For determination of the value of $T^{(s)}$ it is necessary to calculate the nonproductive expenditures of DUK time in each mode.

The nonproductive expenditures of DUK time in the time intervals t_1 and t_2 under asynchronous conditions are related to the performance of check tests and switching from one EUM to another. The expenditures of DUK time on performing the check tests are $(t_1/T_{per})\tau_{per}$ and $(t_2/T_{per})\tau_{per}$, and the expenditures connected with switching amount to $\Lambda_0(t_1+t_2)t_{res}$, where t_1/T_{per} and t_2/T_{per} are the number of performances of check tests in the corresponding time intervals, and $\Lambda_0(t_1+t_2)$ is the average number of switchings with the simplest flow of EUM failures. Therefore the value of $T^{(1)}$ can be defined as:

$$T^{(1)}=t_1-\frac{t_1}{T_{\underset{(1)}{n}}}\tau_n+t_2-\frac{t_2}{T_n}\tau_n-\Lambda_0(t_1+t_2)t_{\underset{(2)}{p}}. \qquad (5)$$

Key: 1. per; 2. res

In the synchronous mode the nonproductive expenditures of DUK time are connected with comparison of the results of the EUM-A and the EUM-B (in the time interval t_1) and the performance of check tests (in the time interval t_2). These expenditures are defined as $(t_1/T_{comp})\tau_{comp}$ and $(t_2/T_{per})\tau_{per}$, respectively. The nonproductive expenditures of DUK time on performing check tests at the times of noncomparison of the results of the EUM-A and the EUM-B are $2\Lambda_0 t_1\tau_{per}$. Consequently, we have

$$T^{(2)}=t_1-\frac{t_1}{T_{\underset{(1)}{c}}}\tau_c+t_2-\frac{t_2}{T_{\underset{(2)}{n}}}\tau_n-2\Lambda_0 t_1\tau_n. \qquad (6)$$

Key: 1. comp; 2. per

The nonproductive expenditures in the load and source division modes are related only to performance of the check tests. Here, considering that in the time interval t_1 both EUM can be busy with servicing calls, we obtain

$$T^{(3,4)} = 2\left(t_1 - \frac{t_1}{T_{n(1)}}\tau_n\right) + t_2 - \frac{t_2}{T_n}\tau_n . \quad (7)$$

Key: 1. per

Considering (2), (5) and (7), we obtain the expression for determining the use coefficient of the computing capability of the DUK in various modes:

$$\eta^{(1)} = \frac{1}{2}(P_{00} + P_{01} + P_{10})\left(1 - \frac{\tau_n}{T_{n(1)}} - \Lambda_0 t_p\right);$$

$$\eta^{(2)} = \frac{1}{2}\left[P_{00}\left(1 - \frac{\tau_c}{T_c} - 2\Lambda_0 \tau_n\right) + (P_{01} + P_{10})\left(1 - \frac{\tau_n}{T_n}\right)\right];$$

$$\eta^{(3,4)} = \frac{1}{2}(2P_{00} + P_{01} + P_{10})\left(1 - \frac{\tau_n}{T_n}\right).$$

Key: 1. per

The discussed comparison method and proposed indices permit selection of the best operating mode of the DUK and at the same time improve the efficiency of using EUM in switching centers.

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INFLUENCE OF THE TRAFFIC CONTROL ALGORITHM ON THE TRANSMISSION CHARACTERISTICS OF MULTIPACKET COMMUNICATIONS

[Article by A. V. Andrianov, Leningrad, pp 9-12]

A study is made of the data transmission network (DT) with packet switching (PS), in which the traffic is controlled individually for each pair of interacting subscribers in accordance with the "window" algorithm [1]. The essence of this traffic control method consists in the fact that the sender subscriber is forbidden to input more than the allowed number W (the size of the "window") of unconfirmed packets into the network. Each packet taken out of the network is confirmed by the receiver subscriber. Therefore the actual maximum packet input rate to the network is determined by the rate of removal of packets by the receiver.

Then the analytical method is used to investigate the following characteristics of the traffic control algorithm: the distribution of the length of the packet queue at the network output (in the queue before the output subscriber line) and the effective rate R of data transmission between the two subscribers. The investigation is performed for the case where all of the packet transmission times -- over the subscriber lines and over the network -- are random variables.

Let a multipacket communication be transmitted between two subscribers connected to the PS network. The law of arrival of packets from the AP_{inp} and also the process of servicing the packets in the line to the AP_{out} are assumed to be Poisson with intensities of λ and μ_2, respectively. The packet delay or confirmation time in the network is assumed to be distributed with respect to an arbitrary law with mean values of t_{c1} and t_{c2}, respectively.

The packet transmission process over a logical connection, considering control, is simulated by a three-phase queueing system.

The first phase -- a W-line queueing system without a queue -- simulates the transmission process of data packets of the investigated logical connection over the network. The equipment busy time corresponds to the random packet delay time in the network.

The second phase simulates waiting of the packets in the queue and transmission over the output subscriber line. The second phase is a single-line queueing system with limited queue. The third phase simulates the process of transmitting return confirmations over the network. Its structure is analogous to the structure of the first phase.

The restriction imposed by the traffic control algorithm consists in the fact that the total number i+j+k of packets and confirmations in all three service phases cannot be greater than W. For equality of i+j+k=W, the input traffic halts.

If we denote the steady probabilities of states P_{ijk} where there are i, j and k packets in phases I, II and III, then the system of equations relating P_{ijk} under the conditions $i+j+k<W$, $\mu_1=1/t_{c1}$, $\mu_3=1/t_{c2}$, is written as follows:

$$(\lambda+i\mu_1+\mu_2+k\mu_3)P_{ijk}=(k+1)\mu_3 P_{i,j,k+1}+\lambda P_{i-1,j,k}+ \\ +(i+1)\mu_1 P_{i+1,j-1,k}+\mu_2 P_{i,j+1,k-1}. \quad (1)$$

For limiting states in which i+j+k=W or any of the indices equal to 0, obvious simplifications are introduced into equation (1).

The solution of the system of equations (1) has the form

$$P_{ijk}=\frac{\rho_1^i\rho_2^j\rho_3^k}{i!\,k!}P_{000}\,, \quad (2)$$

where $\rho_1=\lambda t_{c1}$, $\rho_3=\lambda t_{c2}$, $\rho_2=\lambda/\mu_2$, and the probability P_{000} is determined from the normalization condition

$$\sum P_{ijk}=1. \quad (3)$$

From (2) and (3) it is easy to find that

$$P_{000}=1\Big/\sum_{n=0}^{W}\frac{(\rho_1+\rho_3)^n}{n!}\cdot\frac{1-\rho_2^{W-n+1}}{1-\rho_2}. \quad (4)$$

Then the characteristics of interest to us can be defined. The distribution P_j of the queue length before the output subscriber line

$$P_j=P_{000}\cdot\rho_2^j\cdot\sum_{n=0}^{W-j}\frac{(\rho_1+\rho_3)^n}{n!}. \quad (5)$$

Expression (5) permits determination of the volume of the storage element in the output node of the network required for organization of one or several logical connections.

Then some results are presented from calculations by the obtained formulas.

Figure 1 shows the distributions of the number of packets at the network output as a function of the carrying capacity of the output line for $\lambda=1$, W=8, $t_{c1}=t_{c2}=t_c=5$.

From the graphs it is obvious that significant queues of packets can accumulate at the network output not only for $\rho_2>1$, but also in the case of close input and output rates ($\rho_2=1$).

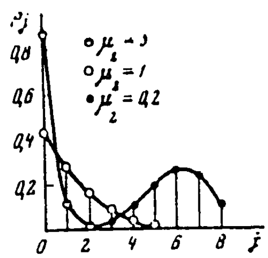

Figure 1

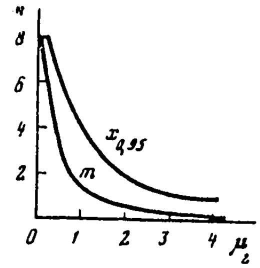

Figure 2

Figure 2 shows the mathematical expectation m of the number of packets j at the network output and the quantile $x_{0.95}$ of the distribution P_j as a function of $\mu_2(\lambda=1, W=8, t_c=5)$.

For illustration of the influence of the parameter W on the efficiency of the control algorithm Figure 3 shows the coefficient $R=\mu_2(1-P_c)/\min(\lambda, \mu_2)$ as a function of the window size W for $\lambda=1$, $t_c=5$ and different μ_2. The relative effective rate R as a function of μ_2 is shown in Figure 4.

It is obvious that the function $R(\mu_2)$ has an extremal nature. The least value of R and, consequently, the greatest losses in speed, as the graphs show, are obtained for $\mu_2=\lambda(\rho_2=1)$. The minimum value of R_{min} decreases with an increase in the network delay.

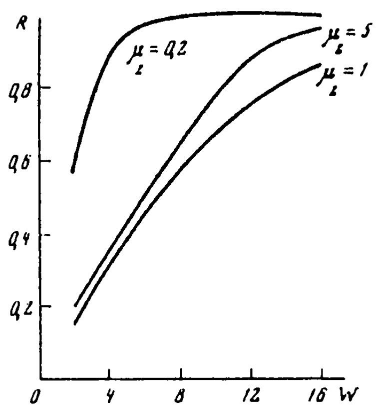

Figure 3

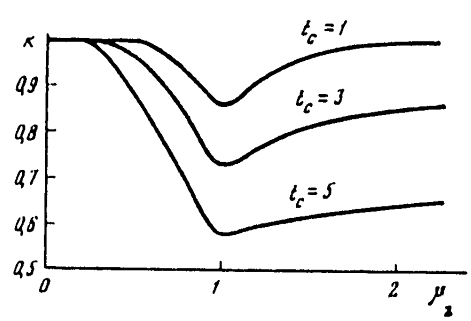

Figure 4

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ESTIMATING THE EFFICIENCY OF A MODEL OF THE GAME METHOD OF DYNAMIC CONTROL OF A COMMUNICATIONS NETWORK

[Article by P. V. Bershteyn and P. I. Kozhemyakin, Kiev, pp 12-15]

A study is made of a network with channel switching in which the intensity of the individual items is insignificant by comparison with the total load serviced by each branch, and therefore the distributed load of the k-th item in practice has no influence on the losses in the branches and, consequently, on the paths by which it is serviced.

A comparative estimate of the application of dynamic control in the above-investigated network using a game automaton of the D type [1] and a version of it [2] is presented in this paper.

Let the flow of the k-th item from some node j be able to be directed over one of the n allowed paths, each of which is characterized by a value of q_i $(i=1, 2,\dots,n)$ called the probability of arriving, that is, $q_i=1-p_i$, where p_i is the probability of failure along the path μ_i. In this case the probability of establishing connection of the calls of the k-th item will be determined by the value of q_k of the selected path. If the calls are distributed with respect to all possible paths using the automaton D, the probability of directing the incoming call to the i-th path will be proportional to q_i. Consequently, the probability of getting through for the calls of the k-th flow using distribution by the algorithm of the automaton D will be:

$$q_D=\sum_{i=1}^{n}\frac{q_i}{\sum_{i=1}^{n}q_i}q_i=\underset{(1)}{q_{cp}}+\frac{\sigma_q^2}{q_{cp}}, \qquad (1)$$

Key: 1. mean

where: $\underset{(1)}{q_{cp}}=\frac{1}{n}\sum_{i=1}^{n}q_i.$

Key: 1. mean

The gain by comparison with using one path will be:

$$B_D=q_D-q_k=q_{cp}-q_k+\frac{\sigma_q^2}{q_{cp}}=\underset{(1)}{\Delta q}+\frac{\sigma_q^2}{q_{cp}}. \qquad (2)$$

Key: 1. mean

If Δq is distributed with respect to a law with probability density $p=f(\Delta q)$, the mathematical expectation of gain

$$\bar{B}_D = \int_{\Delta q_{min}}^{\Delta q_{max}} \left(\Delta q + \frac{\sigma_q^2}{q_{cp}}\right) f(\Delta q)\, d(\Delta q) = \frac{\sigma_q^2}{q_{cp}}. \quad (3)$$

Key: 1. mean

The percentage of failures (losses) is a normalized parameter. It is known that such parameters are distributed most frequently with respect to a normal law [3]. Since Δq is a linear function of p, also Δq is distributed by a normal law with dispersion σ^2_q and mathematical expectation $\Delta q=0$. In this case the probability that $B>0$ can be defined by the formula:

$$P(B_D > 0) = 0{,}5\left[1 + \Phi\left(\frac{\sigma_q}{q_{cp}}\right)\right]. \quad (4)$$

For estimation of the values obtained, it is possible to assume with acceptable accuracy for practice that:

$$\sigma_q = \frac{q_{max} - q_{min}}{6} = \frac{h}{6}. \quad (5)$$

On a well-operating network $h=(2 \text{ to } 3)\cdot 10^{-2}$ and $q_{mean}=1\cdot 10^{-2}$; then

$$\bar{B}_D \simeq \frac{0{,}03^2}{6^2\cdot 1} \simeq 2{,}5\cdot 10^{-5} = 0{,}0025\%,$$

$$P(B_D > 0) = 0{,}552.$$

It is interesting that even for $q_{max}-q_{min}=1$ and $q_{mean}=0.5$

$B_D=5.6\%$ for $P(B_D>0)=0.566$.

It must be noted that when using automata D, almost uniform load distribution with respect to the allowed paths (q_i are almost identical for them) is achieved by the above-described algorithm, although the gain is small.

In order to increase the gain, it is possible to propose an algorithm analogous to the one investigated above except that the call is directed to one of the possible paths with probability proportional to $\gamma_i=q_i-\alpha$. Here $\alpha \leq q_{min}$ is a constant. In this case

$$q_{D\pi} = \sum_{i=1}^{n} \frac{\gamma_i}{\sum_{i=1}^{n}(q_i-\alpha)} q_i = q_{cp} + \frac{\sigma_q^2}{q_{cp}-\alpha}. \quad (6)$$

Key: 1. mean

As is obvious from (6), $q_{D\pi}$ increases with an increase in α.

For $\alpha=q_{min}$ the path with a value of $q_i=q_{min}$ does not participate in servicing the calls of the k-th item. Successively increasing α, we arrive at the type M automaton in which the incoming call is directed to the path with the value $q_i=q_{max}$. Then

$$B_M = q_{max} - q_k. \tag{7}$$

The mathematical expectation of gain will be

$$\bar{B}_M = \int_{q_{min}}^{q_{max}} (q_{max} - q_k) f(q) dq = q_{max} - q_{cp}. \tag{8}$$

In order to obtain a normal probability distribution law for failures with respect to different paths with $q_{max}-q_{min}=h$, for the above-investigated conditions:

$$\bar{B}_M = \frac{h}{2} \simeq (1 \text{ to } 1{,}5)\%,$$

hence

$$\frac{\bar{B}}{\bar{B}_D} = \frac{h \cdot 36}{2 \cdot h^2} = 600.$$

Thus, in channel switching networks with quasisteady probability of losses on the branches, (that is, not depending in practice on the distributed item) the game method of dynamic control using the algorithm of the automaton M turns out to be more effective than automaton D.

The gain from using the automaton M is proportional to the dispersion range of the probabilities of getting through (q_i) along the paths, and for the automaton D, the dispersion of this value. However, with sufficiently high probability (no less than 0.43) the use of automaton D in the above-investigated form also has a negative effect on the service quality.

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ESTIMATING THE RELIABILITY OF COMMUNICATIONS NETWORKS WITH NONIDEAL CONTROL

[Article by V. A. Bogatyrev, Yu. M. Martynov, Moscow, pp 15-18]

By communications network reliability we mean the probability of the presence of at least one properly operating allowable path between a given pair of junctions.

We shall consider that in a communications network with ideal control all possible paths between each pair of junctions are permissible.

In networks with nonideal control situations exist where there is a path through the operating network elements, but it cannot be used for information transmission as a result of absence of information at each communications center on the state of all of the network elements, inaccessibility of individual areas or communication channels for the given data transmission routing, and so on.

Let us propose that as a result of analyzing a specific control method, all of the allowable paths $\mu_1, \mu_2, \ldots, \mu_n$ between each of the given pairs of centers can be enumerated. Each path is given by listing the network elements entering into it:

$$\mu_i = \{a, b \ldots l\}.$$

The path elements can be communication lines and centers. The probability of the existence of each k-th path element is known and equal to p_k. Failures of the communication network elements will be considered independent events. The reliability of the i-th path will be

$$R_i = p_a p_b \cdots p_l. \tag{1}$$

If the given set of permissible information transmission paths consists of independent paths $\{\mu_s, \mu_t, \ldots, \mu_z\}$, the reliability H of the corresponding information routing can be defined by the known parallel connection formula

$$H = 1-(1-R_s)(1-R_t)\ldots(1-R_z). \tag{2}$$

In the general case, however, different paths can contain common elements, and therefore the failures of these paths will be correlated. In reference [1] it is demonstrated that the formula (2) can also be used in the general case, but when the parentheses are removed it is necessary to use the rule

$$p_i p_i = p_i. \tag{3}$$

The necessity for following rule (3) forces the calculations to be performed by formula (2) in general (literal) form; here multiplication by each next binomial doubles the number of terms in the general expression for H so that in the presence of N paths the number of terms in the general formula can reach 2^N.

A method is proposed for significant reduction of the volume of calculations by formula (2) by using a number of simplifying procedures and relations following from (3). Let us denote the multiplication operation subject to rule (3) by the symbol (*). Let us introduce the unreliability index of the i-th element $\bar{p}_i=1-p_i$. It is easy to see that

$$p_i * \bar{p}_i = p_i * (1-p_i) = p_i - p_i * p_i = p_i - p_i = 0. \tag{4}$$

Analogously, it is easy to check the correctness of the following expressions:

$$\bar{p}_i * \bar{p}_i = \bar{p}_i, \qquad 1 - p_a p_b - p_a \bar{p}_b = \bar{p}_a. \tag{5}$$

For illustration of the sequence of the calculations by the proposed method let us consider a simple communication network consisting of eight elements: a, b, c, d, e, f, g, h. Let the control algorithm permit use of only the following paths, the number of elements in which will not exceed three

$$\mu_1=\{a,b\};\ \mu_2=\{c,d\};\ \mu_3=\{e,f\};\ \mu_4=\{a,g,d\};\ \mu_5=\{e,h,d\}.$$

We shall perform the calculation with respect to the probability Q of unconnectedness of the network Q=1-H by the recurrent formula following from (2):

$$Q_t = Q_{t-1} * (1-R_t) = Q_{t-1} - R_t * Q_{t-1};\ Q_0 = 1. \tag{6}$$

Inasmuch as the first three paths in the investigated example are independent (and in the general case the independent paths must first be isolated and put in the first places in the list of paths), for Q_3 according to (2), we have (in order to abbreviate the notation we shall arbitrarily stipulate that $p_a=a$, $1-p_a=\bar{a}$, $1=p_a p_b=\overline{ab}$, and so on)

$$Q_3 = \overline{ab}\,\overline{cd}\,\overline{ef}.$$

Now let us define Q_4 using expression (6):

$$Q_4 = Q_3 - agd * (\overline{ab}\,\overline{cd}\,\overline{ef}) = Q_3 - agd\,\bar{b}\,\bar{c}\,\overline{ef}.$$

According to rule (3) the repeated elements are dropped when the parentheses are removed.

Analogously, for Q_5 we obtain

$$Q_5 = Q_4 - ehd * Q_4 = \overline{ab}\,\overline{cd}\,\overline{ef} - agd\bar{b}\bar{c}\,\overline{ef} - ehd(\overline{ab}\,\bar{c}\bar{f} - ag\bar{b}\bar{c}\bar{f}).$$

In the example of the expression for Q_5 it is possible to formulate the following general algorithm for calculating the probability of disconnectedness of the network by the proposed method. First, the probability of the absence of all independent paths is described by the parallel connection formula, then the probability of the existence of the next path in the list is written with a minus sign multiplied by the preceding notation in which only the probability of the existence of all elements entering into the last path are omitted; then the probability of the existence of the next path is written with the minus sign multiplied by the entire previously obtained expression in which the elements entering into the last path are omitted, and so on.

Let a sixth path of four elements $\mu_6=\{a, g, h, f\}$ also be permissible in addition to the listed ones.

Using the presented algorithm and the expression for Q_5, we obtain the following formula for calculating the network reliability with six allowed paths

$$Q_6 = Q_5 - aghf(\bar{b}\,\overline{cd}\,\bar{e} - d\bar{b}\bar{c}\bar{e}).$$

When calculating the expression for Q_6 we used the fact that $f \times \bar{f}=0$. The cofactor in parentheses, according to (5) permits further simplification

$$\bar{b}\,\overline{cd}\,\bar{e} - d\bar{b}\bar{c}\bar{e} = \bar{b}\bar{e}(\overline{cd} - d\bar{c}) = \bar{b}\bar{e}\bar{d}.$$

Thus, for Q_6 we finally have:

$$Q_6 = \overline{ab}\,\overline{cd}\,\overline{ef} - agd\bar{b}\bar{c}\,\overline{ef} - ehd\bar{c}\bar{f}(\overline{ab} - ag\bar{b}) - \bar{b}\bar{e}\bar{d}aghf.$$

Let us note that when calculating by the generally accepted procedure the expression for Q_6 would contain $2^6=64$ terms.

For demonstration of the possibilities of the proposed procedure let us calculate the increment ΔH of the reliability which takes place if we permit the use of another path $\mu_7=\{e, h, g, b\}$.

Comparing R_7=ehgb term by term with the expression for Q_6, we obtain the addition from the first term equal to $ehgb\overline{acdf}$, and the addition from the third term equal to $ehgbd\overline{cfa}$; the remaining addition terms are not given according to rule (4).

Thus, the desired increment will be

$$\Delta H_7 = ehgb(\bar{a}\,\overline{cd}\,\bar{f} - d\bar{c}\bar{f}\bar{a}) = ehgb\bar{a}\bar{f}\bar{d}.$$

Thus, if the reliability of each element is 0.9, then the increment will be $0.9^4(1-0.9)^3=6.5\cdot 10^{-4}$.

Estimates of this type can be useful when determining the expediency of complicating the communication network control algorithms accompanied by expansion of the subset of allowable information transmission paths. For estimation of the reliability of real communications networks, the discussed method was programmed on a computer; the entire logical part of the calculations are performed using the built-in boolean functions on the rows of bits position-wise representing elements of the communications network. The calculation of the reliability of a two-pole network with three allowable information transmission paths on the YeS-1020 computer takes several minutes of machine time.

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ANALYSIS AND SHORT-TERM FORECASTING OF TELEPHONE TRAFFIC

[Article by R. P. Borisova, T. D. Zelentsova and A. P. Pshenichnikov, Moscow, pp 18-21]

For the dynamic design of city telephone exchanges (GTS) one of the labor-intensive problems is predicting the telephone traffic for newly built automatic telephone offices (ATS) and predicting the traffic variation on all the existing communication lines as a result of a change in capacity and structure of the network. Forecasting is realized on the basis of observing the traffic on the existing network and it consists in developing quality characteristics (description of the observed and expected development trends) and quantitative (point or interval) estimates.

The traffic intensity is a random variable, but in addition to the random fluctuations there are stable trends in the variation of the average values of the load intensity with development of the network. The average load intensity from one group of sources to another does not remain constant with development of the network even if the capacities of these groups do not change. Traffic redistribution occurs, but, as observations show, this process has high inertia. The inertia can be explained obviously by the effect of a large number of factors having opposite influence on the investigated process. Under these conditions, from the large number of existing forecasting methods [1, 2], it appears natural to use statistical methods for predicting telephone traffic load.

The statistical forecasting process consists of two steps [3]. The first step is analysis of the data for the observation period and the period of construction of the statistical model. The construction of the model includes the selection and substantiation of the form of the equation describing the dynamics of the process (its determined base) and estimation of the parameters of the equation. In the second step, on the basis of the statistical laws found, the expected value of the predicted attribute is defined.

The traffic was analyzed for a five-year period beginning in 1970 to 1974 at more than 100 ATS and 2000 interjunction communication lines of the Moscow GTS. During the investigated time period, a unique seven-digit subscriber numbering system was used on the network. One, two, three and four-million numbering zones were organized. A study was made of the traffic intensities occurring at the ATS, the intraoffice, intrajunction, on the routings from the ATS to the outgoing message junctions (UIS), from the UIS to the incoming message junctions (UVS), and from the UVS to the ATS.

The stability of the increase in capacity of the Moscow GTS on the level of 130,000 to 140,000 numbers per year, invariability of the macrostructure of the network during the observation period made it possible to use a very simple linear relation to describe the deterministic basis for the process and load variation in time. The equation parameters were found by the least squares method.

For studying the effect of the capacity of individual groups of subscribers on the traffic flows, the reduced capacities of the ATS were calculated

$$N_{red} = N_{nu} + \sum_i N_i(y_i/y_{nu}),$$

where N_{nu} is the number of residence telephones for private use; N_i is the number of sources of the i-th category included in the ATS; y_i is the average intensity of the individual load of the sources of the i-th category; y_{nu} is average intensity of the specific load of the private residence subscribers.

Let us briefly enumerate the basic results of analyzing the load on different communication lines for the investigated observation period.

The intensity of the load occurring on the ATS is Y_i. The all-office peak load hour (PLH) depends on the structural composition of the ATS subscribers. For ATS with a proportion of private residence telephones in the reduced capacity of the ATS, for more than 50% the PLH falls in the evening hours and and less than 40% in the morning. For the majority of the investigated ATS the ratio of the intensity of the load occurring on the ATS in the PLH to the reduced capacity of the ATS $Y_i/N_{red\ i}$ had a trend toward insignificant growth. The increase in the specific load averaged over all of the ATS of the first million numbering zone was 0.5% per year, and the second million zone, 1.2% per year.

The intensity of the intrajunction load -- the total load from the ATS with respect to all ATS of the junction region I, including the ATS_i - is Y_{iI}. Analysis of the variation in time of the proportion of the intrajunction load in the load occurring at the ATS Y_{iI}/Y_i demonstrated that this ratio had a tendency toward reduction, and for the ATS of the first million numbering zone the reduction on the average was 1.7%, and for the ATS of the second million zone, 0.4% per year. a relation was discovered between the proportions of the intrajunction load in the load occurring at the ATS and the specific weight of the capacity of the region in the total capacity of the network.

The intensity of the intraoffice load of the ATS_i is Y_{ii}. Analysis of the variation in time of the proportion of the intraoffice load in the intrajunction load Y_{ii}/Y_{iI} demonstrated that the magnitude of this ratio increased on the average by 1.8% per year.

The intensities of the interoffice traffic flows (from ATS_i to ATS_j of one junction region) is Y_{ij}. A deterministic basis was discovered for the statistical dependence of the ratio of the traffic intensity in the PLH from the ATS_i to the ATS_j to the intrajunction load of the ATS_i on the specific weight of the capacity of the ATS in the capacity of the junction region I, $Y_{ij}/Y_{iI}=f(N_{red\ i}/N_{red\ I})$.

The intensities of the load on the ATS_i-UIS_k lines are Y_{ik} (k=1,2,3,4). Analysis of the dynamics of this load made it possible to trace the process of traffic redistribution with development of the network. In 5 years the capacities of the million numbering zones increased as follows: the first million by 1.11 times, the second million by 1.18 times, the third million by 3.53 times and the fourth, by 3.42 times. Here the specific weight of the capacity of the first and second numbering zones in the total capacity of the network decreased, and the third and fourth zones, increased. The specific weight of the traffic intensity from the ATS_i to the UIS-1 and UIS-2 in the occurring load of the ATS_i with development of the network decreased, and to the UIS-3 and UIS-4, increased. The traffic redistribution process at the Moscow GTS was also observed earlier, for example, during the 1953-1963 period [4]. Considering that the development rates of individual million numbering zones do not remain constant in time, for prediction of the load in the ATS-UIS directions, it is proposed that the dependence of the ratio Y_{ik}/Y_i on the specific weight of the capacity of the zone k in the network capacity be used.

Intensities of Interjunction Traffic Flows (from UIS to UVS). The study of the dynamics of these loads for 2000 interjunction routes demonstrated that their magnitude is influenced not only by the capacities of the junction regions and network capacity, but also the uniform gravitation factor. Thus, under other equal conditions, the load intensities to adjacent junction regions are greater than to remote ones.

Load Intensities on the UVS-ATS Routings. The specific loads incoming from the UVS to the ATS had a tendency towara growth for the majority of routings, and for the ATS of the first million numbering zone the increase was 1.05% per year, and for the second zone, 0.25% per year.

For short-term traffic forecasting in the majority of cases it is justifiable to assume invariability of the model both in the observation section and in the forecasting section. Under this assumption the determination of the expected loads was realized by extrapolation of the deterministic basis for the observed process. In order to obtain the interval forecast, the forecasting confidence intervals were calculated. It is necessary to solve the problem of short-term traffic forecasting annually; therefore all of the forecasting calculations are performed by computer.

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REAL TIME DISPATCHER FOR A PROGRAMMED SUBSCRIBER'S STATION

[Article by Yu. F. Gal', V. I. Golovach, V. M. Sobol', Ye. G. Stalin, Moscow, pp 21-24]

The creation of prospective programmed subscriber's stations (AP) with a wide spectrum of functional capabilities is connected with the development of software to control the AP resources. The most important operating, reliability and cost characteristics of AP depend on the choice of methods of designing the AP software. Let us note that these characteristics have a significant influence on the efficiency of the corresponding data transmission system as a whole.

A method of developing AP software using the UZOR (real-time operations systems assignments control) dispatcher is discussed. Brief consideration is given to the problems connected with implementation of this dispatcher on mini and micro-computers and the specific nature of its application in AP.

The AP that we are talking about is part of communications systems providing two-way "terminal-terminal" communications in the interests of remote subscribers. The activity of remote subscribers is manifested at the input of the AP as an integral load. In order not to reduce the operating efficiency of the network, the AP reaction must lead the message arrival rate over the communication channels. Consequently, the software-hardware system of the AP must satisfy defined time restrictions; in other words, the operation of the communications AP takes place in real time. On the basis of this specific nature, for programming the AP assignments means are required with the use of which the exchange procedures with the message processing problems or operator activity of the investigated AP are synchronized. The UZOR dispatcher also permits programming of multifunctional AP problems. The UZOR dispatcher is the "frame" of the AP program system, and the functional components of the software of the investigated AP are based on this dispatcher.

The dispatcher controls the following AP resources: the operating time of the central computer processor; the ready-access memory of the computer; interrupts from peripheral devices (for example, communication lines, information storage and display devices).

The programs making up the dispatcher correspond to macroinstructions. From the point of view of the AP software developer, these macroinstructions expand the usual set of instructions of the mini or microcomputer making up the AP, permitting algorithms to be written, individual parts of which are executed in parallel in real time and asynchronously. These parallel parts of the algorithm are called processes. The presence of processes is a characteristic feature of the operation of AP in which several information reception and dispatching problems can be solved simultaneously.

The number of such parallel-solved problems can vary during operation of the AP; therefore the dispatcher has means of assigning dynamic processes which appear for execution of a defined problem and disappear after this problem has been solved. The possibility of assigning dynamic processes makes it possible to use the computer memory which is part of the AP especially efficiently, for on completion of the dynamic process the memory allocated for solution of the problem becomes free and can be used to solve other problems.

Not all of the problems solved by the AP have identical importance. For consideration of the relative importance of simultaneously solved problems at the time of appearance (generation) of each process its characteristic, called its priority, is assigned.

In the computer the processes must have the possibility of becoming synchronized with one another at certain points in time. In other words, the programmer must know how to give events occurring in certain processes simultaneously. Usually at such "meeting points" one of the synchronizing processes ("the sender") transmits the conditional information to the other ("the receiver"). Each of these meeting points corresponds to a signal. The nomenclature and number of signals are selected by the developer of the AP software himself.

Very frequently the expected event consists in the execution of a defined problem. Synchronization with the event consisting in completing the dynamic process can be realized by using the cooperation operator provided for this purpose. The cooperation operator is also convenient for "early" curtailment of certain processes required, for example, in the case of AP operation.

The processing of interrupts from the peripheral devices of the AP is carried out using the interrupt waiting macroinstruction. The characteristic features of the interrupt waiting instruction permit a group of like peripheral devices to be programmed (for example, several alphanumeric displays can be serviced by several processes executing the same program segment).

The UZOR dispatcher also includes special devices that simplify checkout of the AP software in real time. The programmer can set conditions under which emergency situations arise -- disturbances in the operation of the system of implemented processes fixed by the dispatcher himself and capable of leading to failures in the operation of the AP. There is also a possibility of defining reactions to emergency situations which permits the consequences of the failures to be rectified.

The following characteristics of the UZOR dispatcher are noted:

The dispatcher instructions are machine-independent; thus, the dispatcher can be executed on a wide range of computers;

It is especially efficient to implement a dispatcher based on single-processor mini and microcomputers with limited ready-access memory size;

The dispatcher is independent of the peripheral devices; the required drivers (input-output processes) must be programmed by the AP developer himself.

At the present time the UZOR dispatcher is implemented on the SM-3 minicomputer and the "Elektronika-60" microcomputer. The total volume of the dispatcher is less than 2 K of 16-bit words. The software for the specific AP was written using the MAKRO-II microassembler and the macrolibrary corresponding to the dispatcher instructions and operators.

The experimental operation of the UZOR dispatcher demonstrated the following:

The expenditures of labor on the development and checkout of the software were reduced sharply;

The AP software developed using the dispatcher satisfies the requirements of the given speed of the reaction to events in the external environment (requests to receive and transmit data) and restrictions on the size of the ready-access memory used;

The AP software developed using the dispatcher has a clear, flexible, simple structure which permits easy adaptation, accompaniment and modification.

ANALYTICAL AND STATISTICAL SIMULATION OF THE TELEPHONE OPERATION SYSTEM OF AN ELECTRONIC SWITCHING CENTER

[Article by B. S. Gol'dshteyn, Leningrad, pp 24-28]

The telephone operation system (TOS) is one of the most important software components of the switching center, and it is an ordered set of software organizing the computing process in real time, that is, at the natural rate of operation of the switching center and the telephone periphery and providing control of all of the resources of the control computer (EUM).

The telephone operation system is divided into the nucleus and periphery. The nucleus of the TOS is the priority queueing system which is the dispatching mechanism of each specific version of the operation system and is based on a two-dimensional strategy with absolute-relative priorities. Thus, let N flows of requests to include N programs exist to which N priorities are placed in correspondence. Let these priorities be distributed in some way with respect to K levels. On each level k (k=1,2,...,K) there are M_k priorities, the requests of which do not interrupt one another. Thus, it is possible to write a priority inthe form of a pair of numbers (k, m). On the k level there will be (k, 1), (k, 2)...,(k, m), ..., (k, M_k) such pairs. Obviously,

$$\sum_{k=1}^{K} M_k = N.$$

In this system the request z(k,m) for a call for calculation of the priority program (k,m) can be either in the waiting phase /ψ/, or in the servicing phase /$\varkappa$/:

Then the rule for calling the priority program (k,m) implementing the investigated strategy with absolutely relative priorities can be described using the following logical expression:

$$\left\{\pi(k,m) = \psi(k,m) \,\&\, \left[\bigcap_{i=1}^{k} \bigcap_{j=1}^{M_i} \overline{\varkappa(i,j)}\right] \,\&\, \left[\bigcap_{i=1}^{k} \bigcap_{j=1}^{\varphi(k,m-1)} \overline{\psi(i,j)}\right]\right\}_t, \qquad (1)$$

where

$$\varphi(a,b) = \begin{cases} M_i, & \text{for } i<a; \\ b, & \text{for } i=a. \end{cases}$$

Expression (1) is understood as follows. The event $\pi(k,m)$ consisting in calling the functional priority program (k,m) takes place when and only when there is a request to start this program, requests to start higher priority programs are absent, and a higher priority program or lower relative priority program, but of the same absolute priority, is not working.

For construction of an analytical model of the TOS [1], the following basic assumptions are introduced:

The input flow of priority requests (k,m) is Poisson with the parameter $\lambda(k,m)$;

The flows of requests of different priorities are independent, and, consequently, the total flow is Poisson with the parameter

$$\sum_{i=1}^{K}\sum_{j=1}^{M_i}\lambda(i,j);$$

The operating times of the functional programs servicing the requests of different priorities are independent random variables with distribution functions $B_{k,m}(t)$, first moments $b(k,m)$ and finite second moments $b^{(2)}(k,m)$;

A study is made of the steady-state operating conditions of the system, that is,

$$\sum_{i=1}^{K}\sum_{j=1}^{M_i}\lambda(i,j)\cdot b(i,j)=\sum_{i=1}^{K}\sum_{j=1}^{M_i}\rho(i,j)<1; \qquad (2)$$

The interrupts and exchange operations connected with them do not increase the time spent by the requests in the EUM, interruption of the service program does not lead to loss of time already expended on servicing;

The sizes of the request queues are selected in such a way as to exclude the possibility of loss of a request as a result of absence of places for waiting.

For the investigated strategy of priority organization of the telephone operation system based on the so-called direct methods [2], analytical expressions are obtained [3] for the average waiting time for the beginning of servicing of a priority request (k,m):

$$V(k,m)=\frac{b(k,m)-\theta(k,m)\sum_{j=1}^{m-1}\rho(k,j)}{1-\sum_{i=1}^{k}\sum_{j=1}^{\varphi(k,m-1)}\rho(i,j)} \qquad (3)$$

and the average total servicing time, that is, the time from the beginning of the work of the service personnel to completion of the servicing of the priority request (k,m)

$$W(k,m)=\frac{\frac{1}{2}\sum_{i=1}^{k}\sum_{j=1}^{\varphi(k,m)}\lambda(i,j)\,b^{(2)}(i,j)+\left[1-\sum_{i=1}^{k-1}\sum_{j=1}^{M_i}\rho(i,j)\right]\sum_{j=m+1}^{M_k}\rho(k,j)\lambda(k,j)}{\left[1-\sum_{i=1}^{k}\sum_{j=1}^{\varphi(k,m-1)}\rho(i,j)\right]\left[1-\sum_{i=1}^{k}\sum_{j=1}^{\varphi(k,m)}\rho(i,j)\right]}, \qquad (4)$$

where

$$\theta(k,m)=\begin{cases} b(k,m), & \text{for } k=1, \\ \dfrac{1}{\sum\limits_{i=1}^{k-1}\sum\limits_{j=1}^{M_i}\lambda(i,j)}\displaystyle\int_0^{\infty}\Big[1-\exp\Big\{-t\sum_{i=1}^{k-1}\sum_{j=1}^{M_i}\lambda(i,j)\Big\}\Big]dB_{k,m}(t), & \text{при } k\neq 1 \end{cases} \tag{5}$$

and

$$\nu(k,j)=\begin{cases} \dfrac{b^{(2)}(k,j)}{2b(k,j)} & \text{for } k=1, \\ \dfrac{b(k,j)-\theta(k,j)}{b(k,j)\sum\limits_{i=1}^{k-1}\sum\limits_{j=1}^{M_i}\lambda(i,j)} & \text{for } k\neq 1. \end{cases} \tag{6}$$

Expressions (3)-(6) permit a priori information to be obtained about the quality indices of the functioning of the TOS and realization of the search for the optimal strategy of the priority servicing system. Considering the labor intensiveness of the calculations by the proposed procedure of complex multipriority system, special programs were written to calculate the TOS parameters in the PL/I language [4].

However, the parameters defined by the presented expressions, just as any average values, are not in the general case sufficiently complete characteristics of the computation process in the EUM of the switching center, the behavior of which is of a probability nature. Therefore the problem of finding more representative characteristics of the TOS, such as the distribution function of the time spent by the requests in the EUM and the queue length distribution functions for the entire set of request flows is very urgent. Unfortunately, the definition of these functions by analytical methods is very difficult, and results usable in practice have not been obtained in this area. For solution of the stated problem the method of statistical simulation of the telephone operation system was selected.

In exactly the same way, the analytical methods do not permit estimation of the operation of the TOS in the overload mode, that is, on violation of the condition of being steady state (2). This situation is reached in the switching center during the periods of high activity of the telephone periphery when the total demand for EUM resources of the set of requests forming the current load of the control computer can exceed the capability of the system. Under these conditions the telephone operation system reveals its true indices. The TOS is studied in the overload mode by the statistical model.

It must also be noted that this model is free of the majority of restrictions characteristic of the analytical method of describing TOS, and it compensates for the difficulties in experimental determination of the probability characteristics of the computation process during operation of the switching center in real time.

What has been stated permits consideration of the problem of constructing a statistical model of the telephone operation system a necessary step in the operations when designing the software for electronic switching centers The given model is a means in the hands of the TOS developers which permits rapid and quite accurate deep analysis of the operation system and makes it possible to obtain a justified estimate of the strategy of organizing the computation process.

The basic thing in the statistical model of the TOS is simulation of the interrupts and functions connected with completion of the operation of the service program and calling another program. This permits variation of the current simulation time in hops, the duration of which corresponds to the free times of the central processor.

Evolution of the computer process consists in interchange of the phases of time spent by requests by the EUM accompanied by modifications of their dynamic parameters. For description of the current state of the s-th request of the ℓ-th priority, the EUM uses the TAB vector (s, ℓ). The components of this vector are the current characteristics of the state of the request: the time of initiation of the request (MOM); the operating time of the functional program servicing the given request (TSA); the real time of servicing the request accumulated in "lots" in the intervals between interrupts of the service program (TSR); the time of waiting for the beginning of operation of the service program (TWB); the time of waiting for continuation of operation of the service program accumulated in "lots" during interrupts (TWI).

The operating process of a telephone operation system in the model is considered during the time period (0, τ), that is, the requests for which the initiation time MOM>τ do not reach the system and are not serviced. The value of the simulation interval τ is selected from the conditions of obtaining sufficiently representative statistical data.

For the selected strategy of priority organization of the TOS using the model it is possible to isolate the limits within which the parameters of the operation graph are located so as to insure predominance of this strategy over the others. In addition, the developed model permits assembly of additional statistical characteristics of the telephone operation system: the average number of requests serviced without waiting; the average number of interrupts of the functional program servicing the requests of ℓ-th priority; and a number of others required to estimate the operating quality of the TOS.

The model is written in PL/I language and is formed as a set of procedures: the procedure for generating the operation graph GEN, the main simulating procedure MOTOS, the procedure for processing the simulation results MINAN and FINAL. This made it possible to create and check out a complex simulating system having quite broad functional possibilities in very short time.

The investigated methods of studying the telephone operation system of the electronic switching center were used when developing a specific version of the TOS of the time-pulse tandem communications junction.

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METHODS OF DYNAMIC DISPATCHING OF PROBLEMS IN COMPUTER NETWORKS

[Article by K. R. Guaryan, V. M. Konovalov, Moscow, pp 28-32]

Large material expenditures on the creation, operation and maintenance of computer networks are giving rise to the necessity for improving the operating efficiency of the computer systems within the frameworks of a united network, the creation of methods and algorithms for operative control of the hardware, software and information reserves of the networks.

The problem dispatching systems known at the present time, as a rule, provide for centralized decision making in the network with respect to the processing center. The low functional reliability of centralized dispatching systems and the necessity for realizing traffic distribution in real time on arrival of a request for processing are promoting the creation of decentralized dynamic dispatch systems (SDD), the development of which must be realized in the set of problems connected with creating network operation systems.

In this article a study is made of the principles of decentralized dispatching in which the dispatcher functions are distributed with respect to all the junctions of the network. The junction SDD makes a selection of the computer when the request arrives for processing. The flows of problems and the time for processing them are random variables.

The methods of decentralized dynamic dispatching are distinguished by the strategy used in the network for gathering information on the state of the computing and communication reserves. When implementing an information strategy I_{S0} in the network, the service information (SI) is gathered in the training mode; the SI accompanies the problems sent through the network and the calculation results. For the strategy I_{S0}, the trained SDD -- SDD(0) -- are proposed which operate under conditions of a priori indeterminacy of the situation in the network, the shortage of SI is made up by the SDD(0) in the operating process. The realization of a developed network metering system in the network corresponds to the information strategy I_{S1}. In the corresponding SDD -- SDD(I) -- a "fast" model of the computer network and its components is used. The efficiency index of the functioning of the junction SDD is assumed to be the average waiting time of the calculation results characterizing both by output capacity of the network and the quality of servicing the users.

The construction of mathematical models of decentralized dispatching for information strategies I_{S0} and I_{S1} is realized using the game-theory methods. Decentralized dynamic dispatching can be represented in the form of collective behavior of N junction SDD, each of which strives to minimize the intrinsic quality index $J_i(y_1,y_2,\ldots,y_N)$ related to the average time the user problems are in the network, where $y_i \in Y_i$, $i=\overline{1,N}$ is the SDD strategy. Inasmuch as the junction SDD have common interests -- the effort to reduce the time expenditures on processing the problems as a whole throughout the network -- the situation that arises is analogous to games of N people with nonzero sum, the optimalness principle in which is the Nash equilibrium. Each junction SDD distributes the problems coming to its own junction between the junctions N_1 in its range. The SDD strategies satisfying the Nash condition are optimal inasmuch as they lead to a stable situation in which the SDD are in no position to decrease the time the problems spend in the network by varying their own strategies.

For information strategy I_{S0}, multi-input stochastic automata are used as the basis for the training SDD. These automata function in random media by which we mean the computational and communication processes occurring in the computer network. The decentralized dispatching reduces to collective behavior of N automata A_i, $i=\overline{1,N}$ in this case:

$$A_i=\{X^{(i)},S^{(i)},Y^{(i)},Q_A^{(i)},P^{(i)},T_L^{(i)}\}.$$

Here X are the input signals (nonbinary alphabet) representing the time the individual problem is in the network $w(t)$; S is the number of states of the automaton, each of which corresponds to a defined computer. The output signals Y are the number of the computer selected by the incoming problem. The output function $Q_A:S\to Y$ is deterministic. The elements of the vector of state of the automaton $P(t)$ define the probability of selecting the next state of the automaton at the next point in time. Adaptation of the SDD is achieved by varying the elements of the vector $P(t)$ in accordance with the Fu-MacLauren training. During the interaction of the junction SDD, the average processing time of the problems in the network for the SDD under the most unfavorable conditions from the point of view of having computer resources is minimized.

For the information strategy I_{S1}, a model of decentralized dynamic dispatching is presented which is formalized within the framework of the differential game theory. Proposing simulation of the computer reserves by one-line Markov queueing systems (SNO), the dynamics of the average problem processing time in the computer network are described by a differential equation of the type

$$\dot{W}=\frac{1}{\lambda}\sum_{i=1}^{N}\lambda_i\dot{W}_i=\frac{1}{\lambda}\sum_{i=1}^{N}\lambda_i\left(\frac{\lambda_i}{\mu C_i}-\frac{W_i\mu C_i}{W_i\mu C_i+1}\right),$$
$$\lambda=\sum_{i=1}^{N}\lambda_i,$$

under the initial conditions $W(t_0)=W_0$. Each SDD(I) is characterized by a set of control elements arranged by selecting the value of $Y_{ij}(t)=\beta_{ij}\lambda_i(t)$; $i,j=\overline{1,N}$ on satisfaction of the conditions $\beta_{ij}(t)\geq 0$, $\sum_{i=1}^{N_1}\beta_{ij}(t)=1$. The efficiency criterion of the junction SDD assumes the form

$$J_i = \frac{1}{\lambda_i} \int_0^T \sum_{j=1}^{N_i} Y_{ij} (W_j + T_{ij})\, dt, \quad i = \overline{1,N},$$

where T_{ij} is the average time for sending the problem and the results of the calculations between the i-th and the j-th junctions; W_j is the average problem processing time on an individual computer. The solution of the differential game of N SDD is realized both analytically and algorithmically.

The investigated models of decentralized dispatching are used as the basis for the dynamic load distribution algorithms operating in real time [1]. For estimation and forecasting of the dynamic characteristics of the computers in the dispatching algorithms, procedures for determining the average waiting time in the queue W_q and average time the problems spend in the SMO (computer) W_s operating under nonsteady conditions are implemented. In particular, the procedure for calculation and prediction of W_s is based on the expression

$$\dot{W}_s(t) = e^{-\mu t_2} [\varrho(t) - \frac{W_s(t)\mu}{W_s(t)\mu + 1}]; \; W_s(0) = W_{s0}.$$

Prediction of W_s for any point in time consists in multiple solution of the above-presented nonlinear differential equation under various initial conditions for the junctions in the range of the SDD.

On transition from the Markov models of computers to systems of the M/G/1 type the basic characteristic of the state of the computer by which the control strategy is formed in the SDD is uncompleted operation of U(t) during the calculation of which the reliability of the user information about the processing time of the problem in the computer and the actual time for fulfillment of the problem are considered. For priority queueing (computer)systems U(t) is determined by calculating the busy interval.

Inasmuch as the network computers function, as a rule, under conditions of variation of intensity of the problem flows, the efficiency of the SDD(I) depends on the time of gathering information about the state of the network reserves. For Markov nature of variation of the load at the computer input, a procedure is presented for calculating the update period for SI on the state of the computer. The asynchronous principle of SI exchange, the period of which is selected directlv in the network junction, is economically most justified.

A comparison of the proposed dispatching methods is made on a specialized simulation model of the computer network. During the course of the experiments it was established that the SDD(O) and SDD(I) insure reduction of the average time the problems are in the network by 1.6 to 5 times by comparison with networks where there is no dispatching. The SDD(I) decrease the value of W by comparison with the SDD(O) by 9-13% on the average, but they require transmissions approximately 3 times greater than the volume of SI.

In order to improve the effect of dispatching, the range of the SDD must be limited. For SDD(0) the number of available computers must not exceed $N_1=5-6$. A comparative analysis of SDD(0) and SDD(I) with respect to efficiency and complexity of realization demonstrated that the range of application of SDD(0) can be networks with limited gathering of SI, random load surges and the necessity for rapid selection of the computer, above all, the network based on mini and microcomputers. The SDD(I) insure higher quality of dispatching in the presence in the network of a network metering service which is characteristic for networks of medium and large computers.

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METHODS OF INSURING OPERATING STABILITY OF A MESSAGE SWITCHING NETWORK

[Article by I. M. Gurevich, V. K. Demin, V. M. Chentsov and S. Ya. Shorgin, Moscow, pp 32-38]

Let G(V,U) be a message switching network; let V be a set of switching centers of the network (for identification of the network subscribers with the network centers); let U be the set of network communication lines, $|V|=n$, $|U|=m$; $\Lambda=||\lambda_{ij}||$ be the matrix of the load presented for servicing where $\lambda(ij)$ is the intensity of the flow of messages requiring transmission from junction i to junction j; $i,j\in V$ and c_{ij} is the cost of one message of the flow $\lambda(ij)$.

Let V_2 be the set of corresponding pairs (i,j), $i,j\in V$ and $V_2=\bigcup_{i=1}^{\ell} V_i$; $V_i\subset V_2$; $\cap V_i V_j=\emptyset$; $i\neq j$; $i,j=\overline{1,\ell}$ is the breakdown of V_2 into priority classes with respect to cost of the messages.

We shall characterize the quality of servicing the network subscribers by norms for maximum admissible delays $\tilde{T}^k_{ij}$ in transmission of messages of given classes of priorities and the corresponding loss norms $\tilde{\pi}_{ij}{}^k$ of the transmitted messages where $k=\overline{1,\ell}$. Let us isolate the two causes of losses of messages: losses of messages at the network entrance as a result of limiting the load admitted to the network and loss of a message (ij) inside the network from limiting the buffered memory of the network junctions.

Let us propose that the channel capacities of the initial network G and the message flow distribution rule with respect to G are such that for $\forall(i,j)$,K $T^k_{ij}\leq\tilde{T}^k_{ij}$ and message losses are absent.

We shall consider that the message switching network is subject to the effects of information and structural disturbances, assuming that the information disturbance $\varepsilon(\Lambda)$ is estimated by the value of $(\sum_{(ij)\in V_k}\sum_{k=1}^{\ell}\alpha_k(\lambda_{ij}{}^*/\lambda_{ij}))\cdot 100\%$, where $\lambda^*_{ij}/\lambda_{ij}$ is the relative increase in the flow intensity λ_{ij} and α_k is aweight coefficient. The structural disturbance $\varepsilon(s)$ is the vector $(n_1,m_1)\cdot 100\%$, where n_1,m_1 are the relative proportions of failed communication centers and lines of the network, respectively.

Let:

$T^k_{ij}(\varepsilon(\Lambda))$, $\tilde{T}^k_{ij}(\varepsilon(\Lambda))$ be the magnitude of the delay, the delay norm for the information disturbance $\varepsilon(\Lambda)$ for the k-th class of messages;

$T^k_{ij}(\varepsilon(s))$, $\tilde{T}^k_{ij}(\varepsilon(s))$ be the magnitude of the delay, the delay norm for the structural disturbance $\varepsilon(s)$ for the k-th class of messages;

$\pi^k_{ij}(\varepsilon(\Lambda))$, $\tilde{\pi}^k_{ij}(\varepsilon(\Lambda))$ be the magnitude of the losses, the loss norm for the information disturbance $\varepsilon(\Lambda)$ for the k-th class of messages;

$\pi^k_{ij}(\varepsilon(s))$, $\tilde{\pi}^k_{ij}(\varepsilon(s))$ be the magnitude of the losses, the loss norm for the structural disturbance $\varepsilon(s)$ for the k-th message class;

($\tilde{T}^k_{ij}(\varepsilon(\Lambda))$, $\tilde{T}^k_{ij}(\varepsilon(s))$, $\tilde{\pi}^k_{ij}(\varepsilon(\Lambda))$, $\tilde{\pi}^k_{ij}(\varepsilon(s))$ be some piecewise-constant functions; $\{\theta\}$ be a class of network control procedures.

We shall consider that the communications system $\langle G,\{\theta\}\rangle$ is stable with respect to information disturbances if for $\forall\varepsilon^*(\Lambda)\geq\varepsilon(\Lambda)\geq 0$, where $\varepsilon^*(\Lambda)$ is the given maximum information disturbance, $E\{\theta_i\}\subset\{\theta\}$ such that

$$T^k_{ij}(\varepsilon(\Lambda))\leq\tilde{T}^k_{ij}(\varepsilon(\Lambda)),$$
$$\pi^k_{ij}(\varepsilon(\Lambda))\leq\tilde{\pi}^k_{ij}(\varepsilon(\Lambda)),$$

and the system is stable with respect to structural disturbances if for $\forall\varepsilon^*(s)\geq\varepsilon(s)\geq 0$ $E\{\theta_j\}\subset\{\theta\}$, where $\varepsilon^*(s)$ is the given maximum structural disturbance such that

$$T^k_{ij}(\varepsilon(s))\leq\tilde{T}^k_{ij}(\varepsilon(s)),$$
$$\pi^k_{ij}(\varepsilon(s))\leq\tilde{\pi}^k_{ij}(\varepsilon(s)).$$

In the general case stability can be insured by two means.

A priori Method. Here the problem of insuring stability is formulated as the problem of selecting the structure of the initial network, its channel capacities and information distribution rules such that any admissible information and structural disturbances do not take the quality of servicing beyond the limits of the given norms.

For consideration of possible information disturbances the channel capacities (c_u, $u\in U$) of the network communication lines are calculated by the matrix $\Lambda^*=||\lambda^*_{ij}||$. As for consideration of the structural disturbances, under the assumption of failures of only the communications lines it is expedient to use the following method of constructing a stable network. Let $\varepsilon^*(s)$ determine the maximum number $m_1<m$ of possible allocatable communication lines, $\{M\}$ be the set of side sections of the graph G of the network, $\{M\}_1$, $\{M\}_2$, ..., $\{M\}_{m-1}$ be the ordering of the set of sections such that $\delta((M)\subset\{M\}_i)=i$ where $\delta(M)$ is the power of the section M. Let us consider the series $\{M\}_1$, $\{M\}_2$, ..., $\{M\}_{m1}$ and a heuristic rule $\Pi\langle\{M\}_1, \{M\}_2, ..., \{M\}_{m1}\rangle$ of definition of the set $u^*\subset U$, where $|U^*|=m_1$. Here the problem of a priori support of stability with structural variation of $\varepsilon^*(s)$ is solved by selecting the channel capacities and the distribution rules of the communications with respect to the network $G(V, (U\setminus U^*)\cup U')$, where U' is the minimum with respect to cost [two pages of the original Russian text missing].

Here the problem of optimal distribution Λ in the communications network is stated as the problem of defining $\bar{P}_0(\Lambda)$ such that

$$\min_{\bar{p}} \Phi(\Lambda, \bar{p}) = \Phi(\Lambda, \bar{p}_0(\Lambda)).$$

In reference [2], a decentralized algorithm is presented for insuring minimum average cost of the messages in the communications network granting the right of predominant servicing to the priority messages of greater cost. It must be noted that although the algorithm [2] is constructed for a communication network without loss of messages, it appears entirely substantiated that in the case of equality of sizes of the buffer memory of all network junctions the algorithm of [2] insures optimalness with respect to the cost criterion considering the limited nature of the buffer memory and message losses.

Inasmuch as the algorithm of [2] does not take into account the requirement on the magnitude of the communications delay, the following generalization of it as a solution of the system is natural

$$\begin{aligned} &\sum_{i,j} C_{ij} \Lambda^*_{ij}(\ell) \to \min_{\lambda_{ij}(\ell)}; \\ &\sum_{v \in \ell} T_v \leq T_{ij}; \\ &T_v = \frac{1}{\mu_v \cdot C_v - \lambda_v^*}; \quad \lambda_v^* = \sum_{i,j} \sum_{\ell \in \mathcal{L}_v} \lambda_{ij}^*(\ell), \quad \lambda^*_{ij}(\ell) \leq \lambda_{ij}(\ell), \end{aligned} \tag{1}$$

where ℓ is the message transmission route; T_V is the message delay in the line V; λ_V is the average flow in the line V of the route; $\lambda_{ij}(\ell)$ is the intensity of the flow of messages arriving at the junction i which must be transmitted to junction j along the path ℓ; $\lambda^*_{ij}(\ell)$ is the actual passed load; μ_V is the average length of message in the line V; c_V is the carrying capacity of the line V; $\mathcal{L}$ is the set of paths.

Under the assumptions of reference [3], the solution of system (1) as a problem with linear purpose function in the piecewise linear restrictions can be found for restriction of the mode $\lambda_{ij}(\ell)$ to the value of $\lambda^*_{ij}(\ell)$ for fixed routes ℓ, simultaneous determination of the routing ℓ and the restrictions $\lambda^*_{ij}(\ell)$.

As for the algorithm [2], it is easy to see that it is easily adapted both to the information and structural disturbances (not exceeding some defined value) by successive recalculations of the routing tables for each pair $(i,j) \in V_2$.

From the general nature of the algorithm [2] and the algorithm described by the system (1), it follows that the given algorithms pertain to the class of algorithms for preventing overloads. It is possible to demonstrate that if the magnitude of the load entering the network does not exceed some critical value, then by the traced degrees of loading of the communication lines the algorithm [2] optimally selects the communications flow distribution plan. In the case of the occurrence of structural and/or information overloads, the servicing characteristics are calculated (the average delay and losses) for the routing rules in accordance with the algorithm [2], and a comparison is made between these characteristics and their given values. From optimalness of the algorithm

[2] it follows that if the values of the calculated characteristics do not correspond to the given service quality, then under the conditions of the given overloads it is impossible to insure the required service quality. In this case insurance of the required service quality for the most important messages with respect to priorities (cost) is insured by limiting the load transmitted to the network.

Let us assume that the set V_2 of corresponding pairs (i,j) is broken down into two subsets: $V_2=V_0 \cup V_1$; the messages reaching the network at the junction i and addressed to the junction j for $(i,j)\in V_0$ have priority in the sense of mandatoriness of the given quality of servicing of the given messages. Here the problem arises of limiting the inflow of nonpriority messages to the network such that under the condition of insuring the required quality of servicing of the priority messages the total losses in the network (both from limiting the load and from limiting the buffer memory) will be minimal.

In reference [2], an algorithm is proposed for limiting the load coming into the network with message switching which agrees with the described optimal routing algorithm (let us note that it is possible to limit ourselves to investigation of the pairs $(i,j)\in V_1$ only; here the load remains fixed for the pairs $(i,j)\in V_0$).

The algorithm operates as follows. First the insurance of the given quality of servicing the messages of class V_0 in the case of absence of messages of the class V_1 by using the algorithm of [2] for $\lambda(i,j)=0$ for $\forall(i,j)\in V_1$ is checked. In the case of guaranteeing the given service quality (otherwise introduction of the reserve resources of the network is required), the admissible magnitude of the additional load from messages of class V_1 is defined which can be serviced while maintaining the given servicing quality of class V_0. This magnitude is determined by the following algorithm. In each step an initial $a_0^{(1)}>0$ is selected (the minimum average cost of the total load transmitted to the network per unit time), it is assumed that $b^{(1)}=0$, and it is proposed that $c^{(1)}$ is the total load from messages of all priorities. Let n=1. In the n-th step of the algorithm, the optimal routing and load limiting algorithm is used for minimum level $a_0^{(n)}$. If the class V_0 message servicing characteristics obtained here do not satisfy the given values, then we set

$$c^{(n+1)}=a_0^{(n)};\ a_0^{(n+1)}=\frac{1}{2}(b^{(n)}+a^{(n)});\ b^{(n+1)}=b^{(n)}$$

and we proceed to the next iteration step. If the service characteristics are satisfactory, that is, $T<\tilde{T}-\varepsilon_1$, $\pi<\tilde{\pi}-\varepsilon_2$, where ε_1, ε_2 are given constants, we set

$$b^{(n+1)}=a_0^{(n)};\ a_0^{(n+1)}=\frac{1}{2}(c^{(n)}+a_0^{(n)});\ c^{(n+1)}=c^{(n)}$$

and we proceed to the next iteration step. If $\tilde{T}-\varepsilon_1\leq T\leq\tilde{T}$, $\tilde{\pi}-\varepsilon_2\leq\pi\leq\tilde{\pi}$, then we stop with this step, selecting $a_0^{(n)}$ for the true value of the minimum cost of messages transmitted to the network.

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METHOD OF IMPROVING THE SERVICE QUALITY ON A CHANNEL SWITCHING NETWORK WITH DYNAMIC CONTROL

[Article by A. Ya. Dolgoselets, N. A. Knyazeva and L. A. Nikityuk, Odessa, pp 38-42]

Constant variation of message flows in the network with time leads to the fact that noncorrespondence arises between the system of directing the flows and the network structure and, consequently, the subscriber servicing quality is negatively affected. The problem of restoring this correspondence can be solved by introducing dynamic traffic control in the network. In the general case with dynamic control in the communications network for estimation of the state of the network or its individual parts the information is accumulated over some period of time and averaged. In accordance with the average values, the traffic distribution plan in the network is varied. The variation of the distribution plan itself can be realized by selecting the corresponding connection paths (basic and bypass).

As the criterion for estimating the traffic distribution plan from the point of view of efficiency of the functioning of the network it is expedient to select the quality of servicing the subscribers. In this case the plan will be considered optimal where the minimum value of the maximum magnitude of the losses among all pairs of corresponding subscribers is reached [1].

In the given paper a method of constructing the traffic distribution plan in the channel switching network with explicit losses and dynamic control by the criterion of guaranteed subscriber servicing quality is proposed. A distinguishing feature of the proposed method is the fact that the process of forming the traffic distribution plan consists in constructing a set of plans and selecting the best of them in the sense of the adopted criterion. Here, the statistical network parameters are calculated in parallel. The selected plan is used on the network during a time interval required for obtaining the next best plan in accordance with the information about the state of the network averaged with respect to the time interval for obtaining the plan adopted for use. Here the guaranteed level of quality of servicing the network subscribers in different time intervals can assume different values depending on the magnitudes of the flows averaged with respect to the preceding time interval.

The initial data when solving the problem of determining the traffic distribution plan are as follows:

1) The structure of the network represented by the connection matrix $B=||\beta_{ij}||$ where each branch β_{ij} is characterized by the capacity c_{ij} -- the number of standard channels forming it -- and also the magnitude of the threshold a_{ij} expressed in the number of channels inaccessible for servicing the excess (dropped from the path of first choice) load;

2) The traffic intensity between each pair of junctions averaged with respect to some time interval $\tau-\lambda_{ij}$;

3) Restrictions on the choice of paths in the form of requirements on the maximum number of tandem connections and given order of succession on the path of the junctions of different category.

The problem of constructing the optimal traffic distribution plan by the servicing quality criterion is formulated as follows.

Let us find

$$P = \min_{\{M\}} \max_{i,j} P^{H}_{ij},$$

where $i,j=\overline{1,n}$, n is the number of network junctions; P^{H}_{ij} is the service quality between the junctions i and j, which depends on the probabilities of losses of the basic and excess loads on the branches of the network and defined as the ratio of the magnitudes of the load lost in the direction i, j and the incoming load; $\{M\}$ is the set of constructed plans.

Let us introduce the following notation: P^{0}_{ij}, P^{u}_{ij}, $P^{cp(0)}_{ij}$, $P^{cp(u)}_{ij}$, the losses of the basic, excess load and the losses of the basic and excess loads averaged with respect to k-plans on the β_{ij} branch, respectively.

The proposed method of improving the subscriber servicing quality consists in generating a converging series of traffic distribution plans with selection and storage in each step of the plan characterized by improved quality, that is, smaller values of max P^{H}_{ij} by comparison with the corresponding characteristic of the stored plan.

For solution of the indicated problem an algorithm is proposed which is the iteration procedure of the following type.

Step 0. The initial values of the loss probabilities in the branches for the basic and excess loads, for example, zero, are given. The iteration number k is taken equal to zero.

Step 1. The matrices of the losses in the branches for the basic and excess loads averaged for the number of iterations are calculated

$$P^{cp(0)}_{ij}(k) = \Big[\sum_{l=0}^{k} P^{0}_{ij}(l)\Big]/(k+1),\; P^{cp(u)}_{ij}(k) = \Big[\sum_{l=0}^{k} P^{u}_{ij}(l)\Big]/(k+1),$$

where $P^{0}_{ij}(l)$, $P^{u}_{ij}(l)$ are the losses in the branches for the basic and excess loads obtained in the l-th iteration.

Step 2. For each source-junction of the network, a "tree" of the first q optimal paths with respect to the adopted weight parameter to all remaining junctions of the network satisfying the requirements on the path organization is constructed. The values of $P_{ij}^{cp(0)}(k)$ and $P_{ij}^{cp(u)}(k)$ of the averaged loss probabilities in the branches of the basic and excess loads are taken as the weight parameters. The superposition of all of the "trees" forms the next traffic distribution plan M(k).

Step 3. The probabilities of load losses are calculated: basic $P_{ij}^{0}(k)$ and excess $P_{ij}^{u}(k)$ on the branches of the network by the method described in [2].

Step 4. The probabilities of load losses with respect to lines for each pair of network junctions $P_{ij}^{H}(k)$ are calculated.

Step 5. If k=0, the obtained plan and its statistical characteristics are stored. If k≠0, the following condition is checked

$$\max_{i,j} P_{ij}^{H}(k) < P,$$

where P is the corresponding quality characteristic of the servicing of the subscribers for the stored plan. If the condition is satisfied, the distribution plan M(k) obtained in the k-th iteration together with the corresponding statistical characteristics is stored. If the obtained and stored plans have identical values of the servicing quality characteristic, it is necessary to compare the probabilities of losses by lines of successive magnitude.

Step 6. The following condition is checked

$$\max_{i,j} |P_{ij}^{H}(k) - P_{ij}^{H}(k-1)| < \varepsilon,$$

where ε is a small number given in advance. If the condition is satisfied, the transition to step 7 is made. Otherwise the iteration number k is increased by one, and transition is made to step 1.

Step 7. End of operation of the algorithm.

Let us demonstrate that the presented iteration process is converging. Actually, since

$$\bar{P}^{cp(0)}(k) = [k \cdot \bar{P}^{cp(0)}(k-1)]/(k+1) + [\bar{P}^{0}(k)]/(k+1) =$$
$$= \bar{P}^{cp(0)}(k-1) + [\bar{P}^{0}(k) - \bar{P}^{cp(0)}(k-1)]/(k+1),$$

we obtain

$$\|\bar{P}^{cp(0)}(k) - \bar{P}^{cp(0)}(k-1)\| \leq [\|\bar{P}^{0}(k) - \bar{P}^{cp(0)}(k-1)\|]/(k+1).$$

The latter expression can be made as small as one might like for sufficiently large k. As a result of the fact that for the same structure of the network the procedure for constructing optimal paths gives similar or coinciding plans M(k) and M(k-1) for sufficiently close values of the average losses in the branches, we find that the matrices $\bar{P}^{0}(k)$ and $\bar{P}^{u}(k)$ are close to the corresponding matrices $\bar{P}^{0}(k-1)$ and $\bar{P}^{u}(k-1)$. In the latter case with sufficiently large k the following condition will be satisfied:

$$\|\bar{P}^{H}(k)-\bar{P}^{H}(k-1)\|<\varepsilon,$$

where $||A||$ denotes any matrix norm. From the last inequality, satisfaction of the following inequality comes directly:

$$\max_{i,j}|P^{H}_{ij}(k)-P^{H}_{ij}(k-1)|<\varepsilon.$$

Analogously, it is possible to demonstrate the correctness of the corresponding estimates for the matrices $\bar{P}^{u}(k)$ and $\bar{P}^{cp(u)}(k)$.

The rate of convergence of the iteration process depends on the closeness of the elements of the matrices $\bar{P}^{cp}(k)$ and $\bar{P}(k)$, that is, the choice of the initial values of the loss probabilities of the basic and excess loads on the network branches.

The use of the loss probabilities of the basic and excess loads on the network branches averaged for the number of iterations has a smoothing effect on the oscillatory phenomenon of the process and, consequently, increases the convergence rate.

It is obvious that the presented algorithm pertains to the class of directional sorting algorithms for finding the local optimum.

Programs were written for the YeS-1033 computer in PL/I language for the given algorithm. The approximate time spent on obtaining the best plan for the network containing on the order of 30 junctions is about 120 minutes.

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OVERLOAD PROTECTION IN CHANNEL-SWITCHED NETWORKS

[Article by V. M. Dubrovinskiy, Kiev, pp 42-45]

During operation of a network, the intensity of the incoming load to the network can increase. If the increase in intensity exceeds the calculated version of the incoming traffic, the network equipment overloads. Since an equipment overload has a negative effect on the quality of servicing the traffic, when designing the network it is necessary to consider operation of the network with overloads.

Let us consider the diagram depicted in Figure 1.

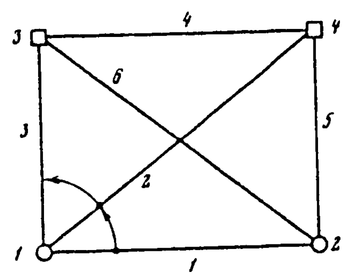

Figure 1

In the given diagram three flows emanate from the junction 1: Y_1, Y_2 and Y_3. Here, the flow Y_1 is under the most favorable conditions, for three groups participate in servicing it: 1, 2 and 3. Flow Y_3 is under the most (sic)* favorable conditions. Only group 3 participates in servicing flow Y_3. The number of channels in this group is determined reckoning that the resultant losses for flow Y_3 will be equal to the given loss norm, that is, $B^{(3)}=B_3^{(3)}=P$. (Here $B_3^{(3)}$ are the individual losses for the flow Y_3 in the group 3.) The resultant losses for flows Y_1 and Y_2 defined by the formulas

$$B^{(1)}=B_1 B_2^{(1)} B_3^{(1)}, \qquad (1)$$

$$B^{(2)}=B_2^{(2)} B_3^{(2)},$$

are found to be less than P here.

*[Translator's note: probably should be least favorabl .]

Accordingly, with an increase in intensity of any flow the flow Y_3 will experience the greatest losses. The flows Y_1 and Y_2 have significant advantages in this case (these advantages are sustained even by comparison with the network without alternative routings where losses for all flows are taken equal to P). Thus, the problem of increasing the stability of networks with bypasses with respect to overloads reduces to protection of the traffic of the last choice group with an increase in intensities of other flows.

At the present time three methods of protecting the traffic of the last choice group are known.

1. Segregation of a separate first choice group for the given traffic (that is, creation of the advantages characteristic of flows Y_1 and Y_2 also for flow Y_3).

2. An increase in the number of channels (within the limits of economy obtained from introducing bypasses) in the last choice group.

3. Use of dynamic control means with restriction of the load thrown in the last choice group.

When using the first method on routings forming last choice paths, two groups are organized: the first choice group (N_c) designed to service the route load itself; the last choice group.

The order of throwing the load to the last choice group for the investigated system depicted in Figure 1 is illustrated in Figure 2 for this case.

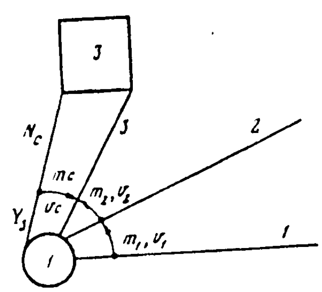

Figure 2

In the given case the resultant losses for the flow itself will be determined by the formula $B^{(3)}=B_c\ B_3^{(3)}$. Here the flow will be protected no worse than flow Y_2.

The number of channels in the third route groups in this case will be calculated as follows. For satisfaction of the required quality of servicing the flows Y_1, Y_2, and Y_3 it is necessary that the losses in the last choice group 3 be selected beginning with the following conditions following from (1):

$$B_{\pi 3} \leqslant \frac{PV_3}{M_3} \frac{Y_3}{U_c};$$

$$B_{\pi 3} \leqslant \frac{PV_3}{M_3} \frac{m_2^{(2)} V_2}{v_2^{(2)} m_2};$$

$$B_{\pi 3} \leqslant \frac{PV_3}{M_3} \frac{m_2^{(1)} V_1 Y_1}{v_2^{(1)} m_2 v_1}.$$

The third relation is always satisfied on satisfaction of the first two.

Since the number of channels in the first and second groups and, consequently, the ratio $m_2^{(2)}V_2/V_2^{(2)}m^2$ are found by definition of N_c, in this case $B_{\pi}3=(PV_3/M_3)(m_2^{(2)}V_2^2/v_2^{(2)}m_2$, and the number of channels N_c must be selected considering the condition

$$\frac{Y_3}{v_c} \geqslant \frac{m_2^{(2)} V_2}{v_2^{(2)} m_2}.$$

By the known N_1, N_2, N_c and $B_{\pi}3$ it is also possible to define the number of channels in group 3.

The investigated procedure permits insurance of almost identical servicing of each of the flows coming into the network without introducing structural redundancy. However, the application of this method encounters difficulties in a number of cases which are connected with complexity of organizing a supplementary routing between the switching offices.

The second method provides for introducing structural redundancy. For calculation of the addition to the last choice groups the necessary initial data are as follows:

The admissible magnitude (k) of the increase in total intensity of flows coming into the system with a common last choice group;

The magnitude of the admissible losses (Bg) for the flow itself and the last choice group with an increase in intensity of any other flow.

For the given values, the addition is calculated in two steps:

The losses are determined for the flow itself under conditions of increasing the intensities of each of the other flows by an amount for which the total intensity of the flows coming into the system will increase by k times, and the ξ-th flow is discovered, an increase in the intensity of which causes the greatest losses for the flow itself;

The addition to the last choice group is determined for which the losses for the flow itself will not exceed the value of B_g under conditions of increasing the intensity of the ξ-th flow to the value obtained in step I. In contrast to procedure I for increasing the resistance of the network to overloads which only creates identical servicing conditions for all the flows, the second procedure permits effective protection of each flow. Thus, whereas with a simultaneous increase in intensity of several (or all) flows the total intensity of the flows coming into the system increases by no more than k times, the losses of each of the flows in the general case will not exceed B_g.

The third method of protecting the flow of the last choice group itself provides for using dynamic flow control means. At the present time many different methods of dynamic control have been created. One of the methods giving rise to the greatest practical interest is the modified Granzhan method in which the threshold $x \leq N$ is established in the last choice group consisting of N channels. For calls of the excess flows, the given group is considered blocked if the number of busy channels in it $q \geq x$. The application of this method permits efficient protection of the flow itself. However, when using it, difficulties arise connected with selecting the effective value of the threshold taking into account both the sizes of the groups and the situation developed in the network.

It must be noted that the effectiveness of each of the investigated methods of increasing the resistance of the network to overloads has still not been finally established at the present time. This arises primarily from the absence of limits for the admissible worsening of quality of servicing of the flows with overload, the specific statistics of the increase in flow intensity in the networks with bypasses and also the absence of a technical-economic analysis of methods.

PROBLEM OF CHANNEL DISTRIBUTION IN THE DIGITAL DATA TRANSMISSION NETWORK FOR RAILROAD TRANSPORTATION

[Article by Zh. Dusembayeva, Alma-Ata, pp 45-47]

The introduction of automated control systems in railroad transportation has given rise to the development of the existing digital data transmission network (PDI). The number of channels of the switched network has been increased, the channel switching means have been improved, and the volume of automatically transmitted data has been increased.

A change has also taken place in the network structure, for the flow routings have changed. New channel switching centers and stations having large information exchange with the computer center for which nonswitchable channels are required have arisen. As a result, the carrying capacity of the switchable network has been enlarged, and the number of stations serviced by this network has been increased. The information subject to transmission over the network is also differentiated with respect to urgency.

Further development of the communications network for the automated railroad transportation control system will take place by joint use of switchable networks operating by the method of channel switching (KK), message switching (KS) and packet switching (KP). Under these conditions the most important operating index is its reliability which can be placed in correspondence to the probability of connectedness of the graph depicting the investigated network. In the given case the problem of constructing the optimal network with respect to reliability is solved by constructing a graph with a given number of apexes and sides in which the probability of connectedness reaches the maximum value.

When designing a communications network for an automated railroad transportation control system, another approach is selected: namely, the network structure is known in general, and it is only necessary to distribute the communication channels optimally under the condition of satisfying a previously given reliability.

For this purpose let us represent the structure of the communications network as a fully connected nucleus, the nodes of which are the switching centers, and the remaining nodes, which are the subscribers, connected in some way to the nodes of this nucleus. Here it is possible to state the problem of determining the optimal number of nodes of the nucleus.

The probability of a random n-apex graph $G_n(t)$ is defined in reference [1], and the probability that the random graph G(n,m) is connected is defined in [2].

In reference [3], the network is depicted by the graph G_n with probability of existence of any side equal to p.

However, in a real network another characteristic is more important: the probability of the existence of paths between arbitrarily selected junctions A and B which corresponds to the value of $P_{AB}(G_n)$. Analogous characteristics (the probability that the message will get from junction A to junction B) were obtained for the KS network -- $P_{AB}(G_n)$.

In the given case a study is made of an information network operating in the KS and the KP mode, for which the probability that the message will get through P (n,T) is defined. Then for determination of the initial assigning graph, the existing network is analyzed, the gravitation matrix and quality characteristics of the function of this network are found. For optimal channel distribution in the investigated network, the assigning graph corresponding to this network has a curve which will be called redundant.

On this redundant curve it is possible to select one side each, obtaining the graph of the corresponding network insuring the same information flows. Considering the cost index, the longest side of the redundant graph is excluded.

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SWITCHING CONTROL IN RAILROAD TRANSPORTATION LONG DISTANCE TELEPHONE SERVICE

[Article by S. L. Dyufur, Yu. V. Yurkin, Leningrad, pp 47-49]

The railroad transportation long distance service network is separated from the national network. It encompasses almost the entire territory of the Soviet Union and corresponds to the railroad network in its configuration. The necessity for this special network is dictated by the responsible role of railroads as the primary transport system on which high requirements are imposed with respect to insuring large volumes of freight and passenger hauling and train traffic safety.

At the present time the development of the network is taking place in the direction of complete automation of setting up calls. Accordingly, the problem of selecting the network structure, the method of controlling the network and optimal value of the quality indices of the servicing of calls have important significance.

The railroad transportation communications network has the following characteristic features as compared to the national long distance communications network:

The line loads are small and are within the limits of 1 to 15 erlangs;

The network operates with calls on the long distance channels with explicit losses and repeated calling where the probability of call losses has an average value of 0.2 to 0.3. At the same time the probability of load losses is close to zero. The study of subscriber opinions demonstrates that losses in the peak load hour of 20% of the calls are considered to correspond to good quality of service as a result of which no significant reduction in the probability of call losses is planned in the long range forecasting of network development;

The network has predominantly a single-route structure, but the necessity for increasing its viability has raised the question of broad application of alternative routings;

The 10-step switching equipment is primarily used on the network; the crossbar system has been partially introduced. The introduction of quasielectronic and electronic systems is planned in the future.

In the given phase of development of the network it is expedient to use the static control method. As the calculated loss probabilities with respect to calls are reduced and modern switching equipment is introduced in the network, conditions will be created for transition to the dynamic method of network control. An expedient number of bypass routes and the number of branches in each route are being substantiated.

On different network levels there are conditions of application of hierarchical and symmetric methods of constructing the network. The limits of expediency of each of the methods must be defined considering the costs of the channels, the incoming loads and the probabilities of losses with respect to calls. Figure 1 shows graphical relations permitting qualitative estimation of the limits of applicability of the symmetric and hierarchical methods of network construction. The coefficient F is the ratio of the total cost of the channels of hierarchical structure to the total cost of channels of a symmetric network. The argument h is the cost of one channel of the network branches with respect to the channel cost in a high-use group for hierarchical structure of the network. Curve 1 characterizes the case where the load coming to each branch of the network is 3 times less than the load coming to the high-use group; curve 3 is the case where the given loads are as many times greater than the load on the high-use channel group; curve 2 shows all of the incoming loads identical. For $F>1$, the symmetric structure has the economic advantage.

For a real railroad transportation network, families of configurations were discovered during the analysis process, and the costs of each version were determined with respect to consolidated indices $Q=K+\mathit{3}T$, where K are the capital expenditures, $\mathit{3}$ are the operating expenses, T is the return time on investments. In Figure 2 as an example we have the relative cost of each version of the network structure (symmetric and hierarchical methods) as a function of the ratio of the branch lengths d. The study confirms the conclusion previously drawn [1] of expediency of application of the symmetric structure in the majority of cases on the railroad transportation network.

The intensities of the incoming flows are determined, and by measurements over a long period of time it was discovered that a significant increase in them with respect to the established level has low probability. On the other hand, loads caused by damage to part of the branch channels or complete failure of the network branches are possible. Simulation has demonstrated that the symmetric networks tolerate overloads connected with damage to part of the branch channels better than the hierarchical network; in the case of total damage to a branch, the network is in practice blocked as a result of an avalanche increase in the repeated calls. In this case the bypasses must be forbidden.

The published research results provide a basis for assuming that in the case of small loads and high use of the network channels, control methods with limited waiting at the call generating point and with threshold for tandem loading will have an advantage with respect to the strategy of servicing over direct groups.

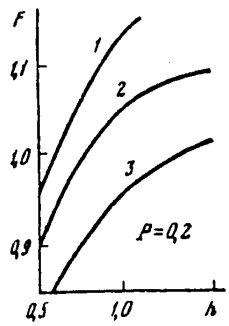

Figure 1

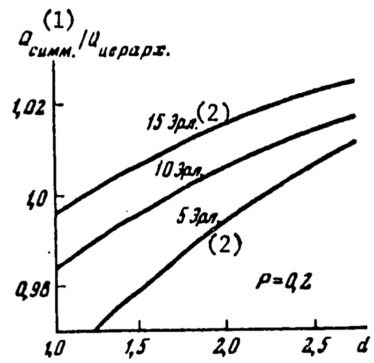

Figure 2

Key:
1. $Q_{symm}/Q_{hierarchy}$
2. erlangs

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CONTROL ALGORITHMS AND CARRYING CAPACITY OF SWITCHING SYSTEMS

[Article by V. A. Yershov, Moscow, pp 49-53]

Effective estimation of the operating quality of automatic switching systems including two functionally complex subsystems -- the control subsystem and switching subsystem itself -- is impossible without joint analysis of them and a systems approach to the given problem. This is manifested especially clearly in prospective automatic switching systems with control based on a computer where the method of processing the control information is determined by the call set-up algorith on which, in turn, depends on the method of finding the connecting paths in the system.

Let us consider the class of path selection algorithms in an arbitrary multilink switching system (KS), which we shall define as follows. Let $G_c^{(i)}$ be a random (equilibrium) selection algorithm for a free segment of the path in the link i and $G_y^{(j)}$ be the ordered selection algorithm for a segment of the path in the link j. Then

$$G=\{G_c^{(i)},\ G_y^{(j)} \mid y\in z_1,\ i\in z_2,\ z_1\cup z_2=z\}, \tag{1}$$

where the z-set of links of KS is a path selection algorithm in KS, for which a random choice is used in the z_2-link, and ordered selection of the path segments, in the z_1-links.

The most detailed study was made of the influence of the algorithm $\{G_c^{(i)} \mid i\in z\}$ representing the random path selection algorithm in the KS, on the carrying capacity. The algorithm $\{G_y^{(j)} \mid j\in z\}$, which is the ordered selection algorithm, has been studied appreciably less in theoretical respects. The algorithms of the more general form defined by (1), were studied only by statistical simulation of the operation of the KS on a computer.

Let us consider the calculation of the capacity of a multilink KS when using algorithms of the type of (1). The basis for the approximate mathematical model is the relations used in the separate loss method (MRP) [1]. According to the MRP, the general loss probability with respect to calls p and its components in the KS operating in the group finding mode (GI) are determined from the conditions:

$$p=p_b+p_v, \tag{2}$$

$$P_b = \frac{(v-d)(1-p)}{YK[1-E_v(\mu_b \frac{Y}{1-p})]} \cdot \frac{E_v(\mu_b \frac{Y}{1-p})}{E_{v-d-1}(\mu_b \frac{Y}{1-p})}, \quad (3)$$

$$P_v = \mu_v E_v(\mu_v \frac{Y}{1-p}), \quad (4)$$

$$K = \frac{K_1 - 1}{K_1},$$

where p_b is the probability of losses with respect to calls as a result of internal blocking in the KS; P_v is the probability of losses with respect to calls as a result of all of the outputs of the hunting route being busy; v is the number of outputs on the hunting route; Y is the intensity of the serviced load on the hunting routing; K_1 is the number of switchboards of the first KS link; μ_b, μ_v are the load screening coefficients assumed to be equal to the following, respectively:

$$\mu_b = 1 - P_v; \quad (5)$$

$$\mu_v = 1 - p + P_v. \quad (6)$$

The magnitude of the effective availability d is determined from the expression:

$$d = (1-\pi)v + \pi\eta Y_1 + \pi(v - \eta Y_1)/K_1, \quad (7)$$

where π is the probability of blocking of the routing output; Y_1 is the load intensity serviced by one switchboard of the first link KS; η is the proportion of the serviced load of the first link switchboard routed on the investigated routing.

For solution of the system of equations (2)-(7), it is necessary to know the load γ serviced bv a group of v lines, the structural parameters of the KS, the interlink cross-connection law given by the graph of the paths between the entrance and exit and the probability π. The value of π depends on the above-indicated parameters and also the algorithm for selecting the path segment in each link.

Let us find the probability π. For this purpose let us consider how the load on the KS links is distributed during random and ordered hunting. Let us introduce the following notation. Let Y_r be the load caused by one switchboard of the link r; $y(r,\ell)(r+1,t)$ be the load serviced by one line connecting the switchboard ℓ of link r to the switchboard t of link r+1, where $r=\overline{1,z-1}$, $\ell=\overline{1,k_r}$, $t=\overline{1,m_r}$. During random hunting in the link r we shall consider that

$$y_{(r,\ell)(r+1,t)} = Y_r / m_r. \quad (8)$$

For ordered hunting in link r, beginning with line s, the distribution $y(r,\ell)(r+1,t)$ will be given as follows. If

$$A_r = Y_r / (1 - E_{m_r}(A_r)), \quad (9)$$

then

$$y_{(r,\ell)(r+1,t)} = A_r\left[E_{t-1}(A_r) - E_t(A_r)\right], \tag{10}$$

where

$$t=\begin{cases} s, s+1,\ldots,m_i,1,\ldots, s-1, & \text{if } s \neq 1 \\ 1,\ldots,m_1, & \text{if } s=1. \end{cases}$$

Let $W(r,\ell)(r+1,t)$ be the probability that the side $R(r,\ell)(r+1,t)$ of the graph of the paths between the entrance and exit of the system is busy. Then, if we assume that

$$W_{(r,\ell)(r+1,t)} = y_{(r,\ell)(r+1,t)}, \tag{11}$$

then the probability graph obtained in this case determines the probability π for the adopted path selection algorithm.

The problem changes insignificantly if KS operates in the GI mode with several attempts to set up a call. In this case, only the expression for the probability p_b changes. It is possible to show that for ν attempts to set up a call the probability of blocking will be:

$$p_b = \frac{\binom{m_z}{d_\nu}}{\binom{n_z}{d_\nu}} \cdot \frac{1}{1-\sum\limits_{x=\nu(n_z-1)+1}^{\nu n_z}[x]_z} \prod_{j=1}^{\nu} \frac{[(m_z-d_\nu)-j+1]}{A\,\frac{K_1-1}{K_1}} \times \tag{12}$$

$$\times \frac{E_{\nu m_z}\left(\Gamma_b \frac{\nu Y_{n_z}}{1-\rho}\right)}{E_{\nu(m_z-d_\nu-1)}\left(\Gamma_b \frac{\nu Y_{n_z}}{1-\rho}\cdot\frac{K_1-1}{K_1}\right)},$$

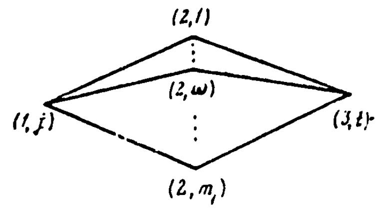
where Y_z is the load caused by a group of n_z lines; n_z, m_z are the number of inputs and outputs of the switchboard of the link z; d_ν is the effective availability of the KS for ν efforts to set up a call; $[x]_z$ is the probability that νn_z lines will be busy simultaneously in ν switchboards of the link z.

Let us illustrate the proposed method of calculating losses in the KS in the example of a three-link KS operating in the GI mode with ordered path finding algorithm. The investigated KS has the following structural parameters: $n_1=m_1=n_3=m_3=k_2=h=18$, $n_2=m_2=k_1=k_3=20$, $\eta=1/18$. For the ordered algorithm the probability of blocking the output of the KS will be defined by the graph, the probabilities of a busy on the sides of which are found from (9) to (11). Considering the configuration of the graph (see the figure) for the probability π we have:

$$\pi = \prod_{\omega=1}^{m_1}\left[1-(1-W_{(1,\ell)(2,\omega)})(1-W_{(2,\omega)(3,t)})\right]. \tag{13}$$

The results of theoretical calculations of the loss probability with respect to MRP (equations (2)-(7), (12)) and the results of statistical simulation for the ordered hunting algorithm and all links of the KS are presented in the table. The results of the statistical simulation are taken from [2].

Y/V	$P_{мрп}$ (a)	$P_{мод}$ (b)
0,603	0,0II5	0,0I20 ± 0,0024
0,702	0,0420	0,0368 ± 0,0038
0,735	0,064I	0,0642 ± 0,0074
0,762	0,0922	0,0824 ± 0,0066
0,786	0,1289	0,II34 ± 0,0079



Key:
a. MRP
b. MOD

Comparing theoretical data with the statistical simulation data, it is possible to note good agreement of them. Some overestimation of the data obtained by the method of separate losses can be explained by the fact that during the calculation the fact that the calculated loads are of a smoothed nature failed to be taken into account [1]. Good agreement of the theoretical data and the statistical simulation data is also obtained in cases where the number of efforts to establish a connection is limited to the value of ν.

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PROBLEMS OF DESIGNING DEVELOPED NETWORKS OF COLLECTIVE-USE COMPUTER CENTERS IN THE DIALOG MODE

[Article by Yu. P. Zaychenko, Kiev, pp 53-57]

Among the problems of designing networks of VTsKP [Collective-Use Computer Centers] and data transmission systems, an important role is played by the problems of topologic design, as a result of which the locations of the VTsKP, concentrators and multiplexers must be determined, the general communications network structure synthesized, and its characteristics defined. The long duration of the period of creation of the VTsKP network and also the variation of a number of initial design parameters in the time interval from development to introduction determine the necessity for transition from traditional static network design problems to dynamic ones.

In the dynamic problem it is necessary to find the plan for development of the VTsKP network and determine the sequence of structures of the developed network for which maximum effect from its use will be insured.

Dynamic Problems of Designing Centralized Networks

Let us consider the statement and mathematical model I of the dynamic problem of designing centralized networks.

Given: the set $X=\{x_j\}$, $j=\overline{1,n}$ is the subscribers of the network -- the sources of problems; the geographic coordinates of each subscriber are $\{\delta_j,w_j\}$; the prediction of the variation in demand for information-computation operations (IV) is $h_j(t)$, $t\in[t_0,T]$, where t_0,T are the times of beginning and ending operations of creating the network; $C_{proc}(t, \Pi)$, $C_{trans}(t,h)$ are the forecasts of variation in cost of information processing and transmission, respectively, with the course of time; the number of steps in creating the network is K; capital investments are W_k allocated for creating the network in the k-th step $k=\overline{1,K}$. It is necessary to determine the plan for network development and find the series of structures $D_1, D_2, \ldots, D_k$ for which the following is insured:

$$\max \sum_{k=1}^{K} H_a(D_k/D_{k-1}) \qquad (1)$$

under the conditions

$$W_\varphi(D_k/D_{k-1}) \leq W_{k\,\text{alloc}}, \quad k=\overline{1,K}, \tag{2}$$

Key: 1. alloc

$$h_j^a(D_k) \leq h_j^{\text{req}}(t_k), \quad j=\overline{1,n}, \tag{3}$$

Key: 1. req

$$H_a(D_k/D_{k-1}) = H_a(D_{k-1}) + \Delta H_a(D_k/D_{k-1}), \tag{4}$$
$$H_a(D_k) = \sum_{j=1}^{n} h_j^a(D_k),$$

where $H_a(D_k/D_{k-1})$ is the total volume of IVR performed by the network in the k-th step; $W_\phi(D_k/D_{k-1})$ are the required capital investments on converting from the D_{k-1} structure to the D_k structure; $h_j^{req}(t_k)$, $h_j^a(D_k)$ are the required and actual automated volume of IVR of the j-th subscriber in the k-th step, respectively, $j=\overline{1,n}$.

In order to solve the dynamic problem (1)-(4) a basic recurrent relation was found, and a calculation algorithm was developed that uses dynamic programming [1].

A more general statement of the dynamic problem (model 2) is also possible, in which the general means of creating the network W_Σ are given, and it is required that they be distributed in the best way between the steps and the series of structures D_1^0, D_1^0, ..., D_k^0 be found for which (1) goes to the minimum under the conditions

$$\sum_{k=1}^{K} W_\varphi(D_k/D_{k-1}) \leq W_\Sigma.$$

A two-level dynamic programming algorithm has been developed for the dynamic model 2: the optimal distribution of means among the steps $\{W_k^0\}$ is found on its upper level, and on the lower level is the optimal distribution within the step between the corresponding VtsKP and centralized communications networks. Thus, an embedded dynamic programming process is used to solve dynamic problem 2.

Dynamic Problems of Centralized Network Design

In contrast to centralized VTsKP networks, distributed networks are characterized by complexity, multiconnectedness of structure, uniqueness of paths of flow distribution. The problems of analysis and synthesis of networks of this class are described in terms of the "multiproduct flows."

Let us consider the dynamic model of the design of the structure of a distributed network.

Given: the matrix of request $H(k)=||h_{ij}(k)||$, $h_{ij}(k)$ is the required magnitude of information exchange among the network junctions i and j in the k-th step; v_{ij} is the relative cost benefit from information exchange between junctions i and j; M(k-1) is the network structure in the (k-1)-st step which is characterized by the presence of effective communication channels $(i,j) \in M(k-1)$, their carrying capacities $d_{ij}(k-1)$ and achieved output capacity of the VTsKP -- $\Pi_i(k-1)$. Required: to find the matched plan for development of all VTsKP and the structure of the intercenter SPD M(k), for which the maximum effect E from using the network in the k-th step is insured with a restriction on the allocated capital investments $W_{k\ alloc}$. The mathematical model of this problem has the form

$$\max E(Ha(k)) = \max \sum_i \sum_j v_{ij} h^a_{ij} (M(k)/M(k-1)) \quad (5)$$

under the conditions

$$h^a_{ij}(k) \leq h^{TP}_{ij}(k), \quad (6)$$

$$\sum_j \gamma_{ij} h^a_{ij}(k) \leq \Pi_i(k), \quad (7)$$

$$W_\Phi(M(k)/M(k-1)) = \sum_{i=1}^{m} C_i^{\text{обр}(3)} (\Pi_i(k)/\Pi_i(k-1)) + \sum_{(r,\ell)\in M(k)} C^{\text{пер}(1)}_{r\ell} (M(k)/M(k-1)) \leq W_{k\ \text{выд.}(2)}, \quad (8)$$

Key: 1. trans; 2. alloc ; 3. process

where $C_i^{process}(\Pi_i(k)/\Pi_i(k-1))$ are the required expenditures for increasing the output capacity of the VTsKP to the magnitude of $\Pi_i(k)$; $C^{trans}_{r\ell}(M(k)/M(k-1))$ are the expenditures on increasing the carrying capacity of the channel (r, ℓ); γ_{ij} is the specific labor intensiveness of handling the traffic h_{ij}.

As a result of analysis of the model (5)-(8) it was established that in contrast to the model (1)-(4), for model (5)-(8) the condition of additiveness is not satisfied. This excludes the possibility of using the method of dynamic programming. Inasmuch as in full, the problem (5)-(8) cannot be solved strictly, the following simplifying assumptions are introduced for its solution:

1) The principle of priority in development of the network is introduced which means that during synthesis of the developed network, the appearance of only those communication lines which must be used also in the final topology of the network is permitted;

2) In each k-th step not all of the centers are developed, but only some subset of them Y(k).

The method of solving the dynamic problem (5)-(8) includes two procedures.

Procedure 1 consists in determining the subset of VTs [computer centers] which must be developed in the k-th step Y(k) and finding the traffic $h^a_{ij}(k)$, transition of which will insure the greatest effect. For this purpose, the exponent $v_{ij}/C^{\gamma\Gamma}$ is used where v_{ij} is the effect from transmission of the traffic h_{ij}, c_{ij} is the sum of the expenditures on transmission and processing of the traffic h_{ij}. Then the perimeter expenditures $W_i(k)$ on development of the i-th VTsKP and the value of $W(k)=\sum_{i=1}^{m} W_i(k)$ are determined.

Procedure 2. The remains of the means for development of the SPD are determined -- W^{trans}_{Σ} -- after which, using the values found for $\{h^a_{ij}(k)\}$, we solve the problem of optimal selection of the carrying capacity (VPS) by the criterion of the minimum C_{Σ}^{trans} or T_{mean}.

The presence of undefined initial data and also several criteria in the network design problems determine the expediency of using the dialog mode. The application of the dialog mode insures the possibility of combining powerful mathematical methods and unformalized knowledge of the designer in a single process. For realization of the dialog mode, a dialog system was developed for topologic optimization of networks DIATOS [2] based on the BESM-6 computer and the UOOGI graphical display.

The system software consists of functional and support subsystems and a control program. The functional subsystems include the optimization subsystems, subsystems for analysis, correction and input-output. The optimization subsystem includes operating programs for optimal location of the VTsKP and KD, optimization of the network topology, and so on; the analysis subsystem permits calculation of the basic operating indices of the network, the correction subsystem offers the possibility of introducing changes into the outgoing and intermediate data and also the dynamic structure using a light pencil.

The interaction between the designer and the system is realized using the system input language.

The DIATOS system permits realization of the interactive mode when solving dynamic VTsKP network design problems, significantly expanding the possibilities of the operating optimization programs. In particular, the following problem is investigated.

Let it be required to determine the optimal plan for the development of a centralized network in model 2 where the capital investments W_Σ are variable, and let it be necessary to find $\max(1/K)\sum_{k=1}^{K} H_a(D_k/D_{k-1}) = \max E_{mean}$ under the condition $W_\Sigma \to \min$.

Both criteria E_{mean} and W_Σ are opposite with respect to effect, and it is necessary to find a compromise solution satisfying both criteria simultaneously. The procedure for finding the compromise solution in the interactive mode consists in the following.

1. Let us determine the range of possible values of W_Σ: $[W_{min}, W_{max}]$.

2. Let us give the initial value of $W_0 = W_{max}$, let us solve problem 2 with respect to one criterion $\max_W E_{mean}(W)$, and find the values of $(E_{mean}(0); W_0)$.

3. Being given the admissible discount ΔW and setting $W_1 = W_0 - \Delta W$, we again solve the problem with respect to one criterion $\max_{W \leq W_1} E_{mean}$, and let us determine the pair of values $\{E_{mean}(1), W_\Sigma(1)\}$.

4. Analyzing these values, the designer makes the decision either to continue the optimization process or curtail it. As a result of repetition of steps 3, 4, a compromise curve is constructed in the plane of the criteria E_{mean} and W_Σ. On this curve, using some additional information (for example, the condition $\partial E_{mean}/\partial W_\Sigma \to \min$), the designer finds the most appropriate point.

On the basis of the proposed algorithms for solving dynamic problems, programs for designing the developed networks of VTs (models 1 and 2) on the BESM-6 have been developed which were used to solve some of the practical problems. In particular, the application of a dynamic model when designing the Latvian SSR network made it possible to increase the average network output capacity by 11% as compared to the static model.

On the whole, realization of the proposed dynamic models permits solution of a theoretically new class of network design problems in which the initial data and parameters are variable, and the use of the interactive mode insures the possibility of considering the multicriterial nature of these problems.

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AN APPROACH TO OPTIMIZING THE STRUCTURE OF A LARGE-SCALE DATA TRANSMISSION NETWORK

[Article by G. P. Zakharov and V. V. Lokhmotko, Leningrad, pp 57-61]

The traditional approach to the solution of the problem of structural optimization of a large-scale PD [data transmission] network on arbitrary weighted graphs turns out to be in practice unrealizable because of the absence of sufficient decomposition experience and limited possibilities of the computer equipment used. A method of automatic calculation of the structure of a large-scale PD network in the class of uniform and regular graphs is proposed which permits significant reduction of the size of the model and realistic optimization of a large network without losses to the basic qualitative properties.

A characteristic feature of the proposed approach which considers a PD network in the form of a set of two subsystems (delivery of messages and technical maintenance) consists in the possibility of coordinating the economic indices of the network, its structural parameters and also the probability-time characteristics of the processes of delivering messages and technical maintenance within the framework of a single mathematical model, which permits recommendation of it not only for selecting the hardware for equipment of the network and topological optimization in the initial design phases, but also when solving the problems of structural network control.

The initial data for the calculation are as follows:

N -- the number of terminal stations of the network (OP) with a breakdown by type;

λ_ξ -- the specific intensity of the outgoing traffic from the OP of the ξ-th type;

ν -- information aging intensity;

V -- volume of message;

ϕ, ψ -- junction and network load closure coefficients reflecting the nature of uniform gravitation between OP;

k_3^k, k_3^y -- the coefficients of the efficiency of use of the communications channel (KS) and switching center (UK) (concentrator);

z_1, z_2 -- dimensions of the rectangle approximating the territory;

p -- types of concentrators and UK distinguished by output capacity $\{G_i\}$, cost $\{c_i\}$ and reliable indices (availability factor $\{k_{ri}\}$ and recovery intensity $\{\alpha_i\}$);

q -- types of KS distinguished by PD speed $\{C_j\}$, the coefficients $\{a_j\}$ and $\{b_j\}$ of the cost function of the line equipment and also the reliability indices $\{k_{rj}\}$ and $\{d_j\}$;

g -- types of technical maintenance centers (TsTO) distinguished by functional accessories and annual expenditures on maintenance $\{e_\gamma\}$.

The dependence of the KS reliability on length $k_{rj}(\ell)$ is proposed.

It is required that the reduced expenditures function П taking into account both capital expenditures of introduction of the network K and operating expenditures З be minimized:

$$П = E_н K + З \rightarrow min \quad (1)$$

on satisfaction of the restrictions on the average time T and probability of timely delivery of messages Q. In particular, it is necessary to determine:

The number of steps in the network hierarchy (R);

The type of structure, types of UK and KS in each step;

Number of UK (n), KS -(m) and TsTO - (h) in each step.

It is proposed that the integral model of the network structure be represented in the form of a set of three functionally distinguished models.

1. Models of the estimate of the probability-time characteristics of the PD network for different switching techniques [1,2,3 and so on], giving the analytical function $T(\lambda, \mu, k_r, d, k_{з0})$ and $Q(\lambda, \mu, \nu, k_r, d, k_{з0})$ where λ and μ are the intensity of the incoming flow and intensity of servicing the UK (KS), respectively.

2. Models of estimating the efficiency of technical maintenance permitting determination of the probability of servicing the next hardware set without waiting $k_{з0}$ [4] as a function of n, k_r, d and h.

3. The topologic model developed by the authors based on the concept of a contour R-separating graph with simple subordination [5] integrating the hierarchical structure by a composition of subnetworks of different levels. Here the spectrum of possible topologies is quantized by a set of base graphs, including the null graph, the shortest connecting network, radial, lattice and uniformly S-connected $2 \leq S \leq n-1$ describing a broad range of structures, from loop to fully connected.

It is proposed that for individual steps in the hierarchy, any of the above-enumerated organizational principles is possible, and for the between-level subnetworks, only radial. For the base topology, analytical relations were obtained between the basic structural parameters such as the number of nodes n, the sides m, the average length of routing π, the average connectedness S, the average geographic length of KS ℓ under the assumption of uniform placement of the OP, and under the assumption of uniform gravitation between the OP of individual subnetworks and realization of routing over the shortest paths, analytical forms of the traffic distribution plan with respect to nodes λ_y and sides λ_k of the network as functions of N, λ_Σ, ϕ, ψ, n, π and S, which in the initial design phases permits avoidance of the application of labor-intensive algorithmic procedures. In the given model optimization problems in the class of fixed structures can be carried out for S_r=const, $r=\overline{1,R}$ (the hierarchical trees, multiloop and radial-nodal structures) and the problems of finding the optimal topology when S_r is unknown.

For the adopted models the purpose function (1) assumes the form

$$\Pi = E_H\Big[\sum_{r=1}^{R}\sum_{i=1}^{p} w_{ir} c_i n_{ir} + \sum_{r=1}^{R}\sum_{j=1}^{q} w_{jr}(a_j + b_j \ell_r) m_{jr}\Big] + \sum_{f=1}^{q} e_f h_f, \qquad (2)$$

$$w_{ir},\ w_r = \begin{cases} 1, & \text{if } i, j \in r \text{ steps}, \\ 0 & \text{- otherwise}, \end{cases}$$

$$\sum_i w_{ir} = \sum_j w_{jr} = 1.$$

The first and second terms in (2) are the capital expenditures on junction and line equipment, respectively, and the third term is the operating expenditures.

The average delivery time on the network is defined as the weighted mean with respect to different routes considering their length π and proportion of the flowing traffic

$$T = \sum_{r=2}^{R} \prod_{i=2}^{r-1} \alpha_i \beta_i \Big[(\varphi_r + \alpha_r \psi_r)\Big(2\sum_{i=2}^{r} T_{i-1,i} + 2\sum_{i=2}^{r-1} T_i + T_r\Big) + \alpha_r \psi_r \pi_r \big(T_{r,r} + T_r\big)\Big]. \qquad (3)$$

Here it is assumed that in the R-step network there are no more than two R routes, that is, through the UK of the r-th step and the subnetwork of the r-th step, $r=\overline{1,R}$, and the delay on the route is represented by the sum of time segments spent by the message in each UK and KS which it visited [6]. Formula (3) is a special case of the expression for the all-network delay [7] for a hierarchical network under the assumption of uniformity of structure within the limits of individual steps of the hierarchy.

The limitation function Q is formed analogously. The difference of this function from (3) lies only in the multiplicativeness of Q for a route taken separately.

$$Q = \sum_{r=2}^{R}\Big[\prod_{i=2}^{r-1} \alpha_i \beta_i Q_i^2 \prod_{i=2}^{r} Q_{i-1,i}^2 \big(\varphi_r Q_r + \alpha_r \psi_r Q_{r,r}^{\pi_r} Q_r^{\pi_r+1}\big)\Big]. \qquad (4)$$

In formulas (3) and (4) the following notation is used:

$T_{i-1,i}$, $Q_{i-1,i}$ are the average time and probability of timely delivery of a message to the KS of an interstep subnetwork, respectively;

T_{r-r}, $Q_{r,r}$ are the average time and probability of timely delivery to the KS of a subnetwork of the r-th step of the hierarchy;

T_r, Q_r are the average time and probability of timely delivery to the UK of the subnetwork of the r-th step of the hierarchy;

$\alpha=1-\phi$, $\beta=1-\psi$.

The procedure for solving the given problem of search for the optimal topology of the network and selection of the optimal type of hardware reduces to substitution of the expressions for ℓ, m, π, T_i, Q_i in (2)-(4) and subsequent solution of the problem of mathematical programming of a combinatory and nonlinear nature. Analysis of the functions (2)-(4) indicates that for fixed R, i, j it can be reduced to a series of simpler functions, the dimensionality of which does not exceed 3(R-1), for in each step, in addition to the subscriber, n, h and S are unknown.

In order to minimize (2) with restrictions (3), (4), a combined algorithm is proposed which is constructed from the ideas of the methods of boundary search and branches and boundaries. Reduction of the sort with respect to the variables R, w_{ir}, w_{jr} is achieved by preliminary cutting out of nonprospective versions based on various asymptotic estimates, quasi and locally optimal solutions.

Program realization of the proposed algorithm was used to calculate the network structures with $N>10^5$, the choice of communications hardware to equip them and discovery of the most effective regions of application of various switching methods.

The results of the studies demonstrated that the proposed approach to optimizing the structure of a large-scale PD network is an effective means of decision making on the part of structural organization of prospective networks.

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TIME DECOMPOSITION OF AN AUTOMATON

[Article by L. N. Zoreva, V. G. Lazarev, Moscow, pp 61-62]

As a result of complication of the operating algorithms of digital automation devices and computer engineering, the necessity for varying the algorithm during the functioning of the digital device, the problem of constructing logical control units (LPU) with adjustable structure combining the advantages of ULU with hardware and software execution of the operating algorithms, is acquiring more and more urgency. This article discusses one of the automatic models of ULU with adjustable structure in the form of time decomposition of the automaton A, and its implementation principle is considered.

Let the automaton A be represented in the form of a composite of subautomata $A_1,\ldots,A_4$. Then the decomposition of the automaton into subautomata $A_1,\ldots,A_4$ will be called a time decomposition if any internal state $\kappa_t^j \in \{\kappa_1^j,\ldots,\kappa_{s_i}^i\}$ of one and only one subautomaton $A_j \in \{A_1,\ldots,A_4\}$ such that the transition function δ_j and output functions λ_j of the subautomaton A_j when it is in the internal state κ_i^j assumes the same values as the transition function δ and output function j of the automaton A when it is in the internal state κ_i, is compared with any internal state $\kappa_1 \in \{\kappa_1,\ldots,\kappa_s\}$ of the automaton A.

Thus, in contrast to the known types of spatial decomposition of the automaton in which each internal state of the automaton A is representable, generally speaking, by internal states of several subautomata, during the time decomposition each internal state of the automaton A is representable by an internal state of one and only one subautomaton A_j.

Obviously, the time decomposition of the automaton, just as the spatial decomposition, can be interpreted as a special encoding of the internal states of the automaton A. Some examples of the coding of the internal states of the automaton A corresponding to the time decomposition of the automaton are considered.

On the basis of determination of the time decomposition of the automaton A, if the automaton A at the time t must be in the internal state $\kappa(t)$, only one of the subautomata $A_j \in \{A_1,\ldots,A_4\}$ will be in the "excited" state, in the internal state $\kappa^j(t)$ which is compared with the internal state $\kappa(t)$ of the automaton A.

It is easy to understand that the time decomposition of the automaton is a breakdown of the set of its internal states into n nonintersecting subsets, with each of which the subautomaton $A_j \in \{A_1,\ldots,A_4\}$ is compared.

One of the criteria of breakdown of the set of internal states of the automaton A into subsets can be the "connectedness" of the internal states by transitions, outputs, and so on.

A discussion is presented of the method of formation of the groups of connected internal states based on the method of obtaining maximum groups of joint internal states of the automaton, and a procedure is presented for constructing the time decomposition of both the general type automaton given in the language of the transition tables and the microprogram automaton given in the language of logical flow charts of the algorithms.

Considering noncomparison of the operation in time of individual subautomata of decomposition structure of the automaton A, each of the subautomata can be realized at different points in time in the same segment of uniform medium of the programmable logical matrix or in another basic module permitting rearrangement (reprogramming) of its structure.

This realization of the time decomposition of the automaton with respect to parts using the same reserves is essentially the hardware-software implementation of the automaton and is a generalization of the known principle of cycle-by-cycle implementation of the automaton.

The structure of the device for hardware-software implementation of an automaton is presented.

DISTRIBUTED CONTROL IN SWITCHING CENTERS USING MICROPROCESSORS

[Article by O. N. Ivanova, Moscow, pp 63-67]

The problems of optimal synthesis of the control units of switching offices and centers of communications networks are an important problem, the solution of which determines the technical-economic indices and expenditures on technical maintenance. In their development the structures of the control units have gone through a number of stages which were characterized by the element base for implementation of the control units (UU), degree of centralization of the UU and switching equipment systems.

In the ten-step ATS* individual relay type control units are used for each switching device. The complexity of the UU was determined by the functions of the switching device in the corresponding finding steps. In the crossbar ATS, relay control units are used for a group of switching devices which form a switching module. The complexity of the UU was determined by the parameters of the switching module and its functions (conditions) in the corresponding finding steps. The transition from individual UU to general UU using the same element base, that is, relays, limited the volume of switching devices which can be serviced by the UU on the basis of low operating speed of the relays (12-30 milliseconds). Therefore each finding stage was constructed from individual switching modules, each of which was assigned a UU. In addition, the volume of switching devices which were serviced by one UU also depended on the operating speed of the switching devices themselves during the process of setting up a call inasmuch as the UU could simultaneously setting up only one call within the limits of the given module. The time of inclusion of the switching elements on the connecting path from the module input to its output was on the order of 50 milliseconds ($t_{mean\ VM}+t_{mean\ UM}$). The effort to increase the degree of centralization of the UU in order to decrease their number could be realized only on a new electronic basis. With the appearance of semiconductor transistors and diodes it became possible to use this element base to construct the UU of automatic telephone offices and centers. This element base came to be called circuit-board electronics, inasmuch as when installing the UU each element was a separate component of the network. The appearance of electronic UU opened up a new area in switching engineering with the building of mechanoelectronic ATS (MYe, Hungarian offices, PS-KE-100, and so on). However, in these ATS systems it was not possible to make full use of the speed of the UU to significantly increase the number of switching devices serviced by one UU inasmuch as the switching devices themselves did not have the necessary speed.

* [automatic telephone office]

The broad possibilities opened up by high-speed electronic UU making use of program control led to finding a new element base for constructing the switching system of the ATS, the basic requirements were good contact quality in the information transmission channel and high speed during the process of setting up and terminating calls. This element base was provided by hercon relays ($t_{response}$=2 milliseconds, $t_{release}$=0.5 millisecond), the ESK relay, ferrides and hesacons. The combination of the switching system constructed from hercon relays or the above-enumerated relays with electron control units led to the development of quasielectronic switching offices and centers. These systems include the ESSI, IOC, IOR, "Kvarts," "Kvant," "Istok" and other systems.

In these systems the UU were constructed on a new electronic base -- integrated microcircuits -- which insured an increase in operating reliability of the UU, simplification of the installation of the UU, and a decrease in size. In the quasielectronic ATS, the degree of centralization of the UU is appreciably higher than for the mechanoelectronic ATS. Especially for the first models of quasielectronic ATS in which all the functions of setting up calls, reception and processing of the address and control information were invested in a control computer (EUM), the output capacity of which would be sufficiently high with high ATS capacity ($250-500 \cdot 10^3$ op/sec for an ATS with a capacity of 10,000 to 20,000 numbers). Thus, at that time the structure of the UU went from the individual UU to the centralized, all-office device which provided control of the switching devices within the limits of the entire switching system (KS) of the ATS. The high degree of centralization of the UU significantly increased the requirements on the operating reliability of the UU of the ATS, for failure of the UU led to shutdown of the entire ATS. Therefore special measures were taken to improve the operating reliability of the UU such as redundancy, deep monitoring and diagnosis of the equipment, which required both increased capital expenditures and increased output capacity of the UU inasmuch as the monitoring and diagnostic programs account for 40 to 60% of all of the instructions executed by the EUM during its operation. After appearance of the quasielectronic ATS with high degree of centralization, deficiencies were discovered in this method in which it is necessary to include the following. With an increase in capacity of the ATS, it is necessary simultaneously to increase the output capacity of the EUM, which leads to the necessity for implementation of it to use the element base with high speed, and consequently, it leads to an increase in cost of the EUM; the requirements on the reliability of the operation of the EUM increase.

For elimination of the first deficiency, the call set-up algorithms were executed in parallel so that part of the processes would take place in parallel, but this, in turn, complicated the general algorithm because it required dispatching when executing individual programs, establishing situations of conflict with simultaneous reference to common modules (for example, the ready-access memory) and introduction of a large number of interrupt levels.

In addition, in order to decrease the influence of the low speed of the KS and terminal sets, part of the EUM functions were transmitted to peripheral UU(PUU) which could interact with the systems and KS without direct participation of the EUM and transmit information to it or receive it on instruction from the EUM.

Thus, the structure of UU with distribution of functions with respect to individual modules of the control units appeared. This method of construction made it possible to lower the requirements on increasing the output capacity of the EUM, because part of its functions began to be performed by individual PUU which can be both passive and active.

In this method of construction, research is required with respect to optimal distribution of the functions among the individual modules of the PUU and the EUM itself and also synthesis of the modules realizing the given function. Then it is necessary to develop the software for the entire UU for the given type of office considering auxiliary types of servicing, automation of technical maintenance and a number of other functions.

Modern progress in the field of electronics and the creation of a new electronic base on the basis of large integrated circuits (BIS) and microprocessors (MP) is opening up new ways to synthesize the UU of switching offices and centers.

Large integrated circuits can carry out quite complicated functional problems during the process of controlling the setting up of a call and can be a device for hardware implementation of the control functions.

In the ATS switching system or the switching system of a center it is possible to isolate operations with respect to setting up calls which are repeated for each call, and they are performed independently, receiving only the instruction to execute and the instruction to output the results of execution from the computer if needed for subsequent operation of the UU. If such operations are encountered in different parts of the system, they can be executed by different BIS, the operating priority of which is given by the operating program of the EUM with respect to setting up the call.

If it is necessary to perform different operations, it is possible to use microprocessors, each of which operates by the corresponding program. Here we have a set of MP, each of which carries out its mission, their operating priority is provided by the EUM which can be executed in the form of a minicomputer with integrated processor which, in turn, can also be executed in the form of an MP.

With this method of construction, the UU will be a set of BIS and MP providing for the execution of all functions of controlling calls at the ATS or center, but each BIS or MP will be connected only to part of the KS equipment or sets, that is, the control will be distributed somehow over individual modules of the switching equipment and the ATS systems. Here the primary principle is the principle of control distribution by which failure of one BIS or MP or another will not disturb the operation of the ATS.

Obviously it is possible to find distributed control structures which will reflect the same structural redundancy as the KS where the connection from the input to the output of the KS or an individual module of it can be provided by several connecting paths from which one is selected. Failure of one path does not lead to disturbance of the possibility of setting up calls but only lowers the quality of servicing the calls somewhat with respect to time required to eliminate failure if it coincides with the PLH [peak load hours]. With this structure, the necessity arises for investigating the capabilities of the manufactured BIS and MP to solve individual control problems at the ATS, to optimize the structure of the UU based on BIS and microprocessor sets and to develop software corresponding to the operating requirements at the ATS or center. The use of distributed control is possible both in the quasielectronic ATS and in the EATS implemented on the basis of digital data transmission. In the digital systems of the ATS, the switching system will be executed by means of electronic contacts which, in turn, can form electronic connectors for the required number of inputs and the number of outputs m realized in the form of one BIS. Such connectors, forming individual modules of the ATS switching system, can control the functional modules of the UU which can be realized by means of the MP operating with the given algorithm for the given switching module. Individual sets (or groups of sets) of one type or another (AK, ShK, SK, RSL, and so on) can also be implemented on the basis of the MP and the BIS.

Inasmuch as at the present time the BIS and the MP are primarily created for the circuitry of computer engineering, it is possible that they do not fully correspond to the requirements which are imposed by the ATS control units; therefore the question can be stated of creating specialized BIS and MP for the UU of the switching centers and offices. For this purpose it is necessary to develop circuitry for the BIS and MP which will to a maximum degree satisfy the requirements on the UU structure.

In the switching equipment the number of BIS and MP required to construct the UU is quite large; therefore the production of specialized BIS and MP in industry can be justified economically. The problem of the developers is to create universal BIS and MP for communications engineering which can be implemented in industry.

The layout of an ATS with a capacity from 2048 to 8192 numbers of the quasielectronic type is considered in which the subscriber finding stage implemented by three-link BAL modules with parameters 2048×1024×512×512, the group finding stages using BSL modules with parameters 512×512×512×512 and the register finding stages using BRI [sic] modules with paramters 512×512×512×512, are used.

For construction of switching system modules, hercon connectors with 8×8 capacitv with electric delay are used. Service wires or OZU [ready-access memory] is used to depict the state of the intermediate lines in the KS.

Control is realized by means of peripheral UU, a central UU implemented in the form of a EUM with program control. The control is constructed so that it is distributed with respect to hierarchical structure. The switching equipment is divided into sections, the equipment in each of which permits setting up calls from the input to the output of the module over independent paths.

In accordance with the breakdown of the switching system into sections, a sectional control system is constructed.

Each section has its own control unit in the form of an identifier of intermediate lines (OPL) and a module for disconnecting the switching elements. The choice of KE is realized by the OPL. The main part of the distributed control system can be implemented on one or two types of microprocessors operating by programs reflecting the operating algorithm of the corresponding module or switching equipment section.

SOFTWARE FOR SUPPLEMENTARY SERVICES OF THE 'KVANT' QUASIELECTRONIC AUTOMATIC TELEPHONE OFFICE

[Article by A. A. Ivanov, Riga, pp 67-71]

The method of controlling the ATS KE [quasielectronic automatic telephone office] by a recorded program gives the subscribers a number of additional services, capabilities or simply additional types of servicing (DVO):

The external communications category;

Constant number within the office;

Determination of the number of the calling subscriber by request of the called subscriber;

Poor audio message;

Extraneous signal message;

Absence of KPV voice-frequency signal message;

Absence of ringing signal message;

Failure of telephone set (TA) message;

Three-way calling or call forwarding;

Speed dialing (up to 30 11-digit numbers);

Direct line (lifting the receiver, without dialing the number);

Temporary inhibition of incoming service;

Temporary selective restriction of incoming service;

Reminder or wake-up calls;

Recording of incoming calls;

Information calls;

Call transfer (from an information call);

Return calling or putting on hold;

Urgent service for privileged subscribers;

Notification outgoing call;

Temporary postponement of completion of DVO;

Conference calling (up to 5 subscribers in one office);

Radio communications;

Search signalling (via a light display);

Dictation machine for recording voice communications;

Automatic telephone answering device;

Group forwarding or night service mode;

Permanent forwarding when the main subscriber is busy;

Recording of outgoing long distance conversations;

Notification incoming call;

Trunk hunting.

The greater part of the listed services are provided by purely software means.

Individual use of the DVO by office subscribers is determined by the class of service of each specific subscriber. The class of service is the set of data which completely describes each subscriber line included in the office, from the point of view of basic and auxiliary forms of equipment and which is stored in memory (ZU), it is compiled when designing the ATS KE by information provided by the requestor, and it can be altered during operation by technical personnel. Organization of the class of service permits any ATS KE subscriber to have any number of DVO in an arbitrary combination.

From the presented DVO list, the first seven are provided to all of the ATS KE subscribers without exception; the last five are realized automatically as a result of the corresponding class of service of the subscriber, and the last DVO are controlled directly by the service user himself by special subscriber procedures performed on the telephone set. Dialog subscriber procedures have been developed on the basis of the Recommendations of the MKKTT [International Telegraph and Telephone Consultative Committee].

Subscriber signalling consists of a set of digital data modules separated by intermodule separators. The first module always contains two digits called the service code.

A unique control signal used in the DVO procedures as the intermodule information separator is dialed by the subscriber by pressing a button ("square on a touch telephone. It is also equivalent to a four-second delay in number dialing, and both signals can be used interchangeably. As a result, identity of subscriber procedures for touch and dial telephones is guaranteed without using any additional nonstandard buttons.

Brief pressing of the microtelephone receiver arm for 0.2 to 1.2 seconds or briefly hanging up (KRO) is used as the signal to get a second line to dial a DVO subscriber procedure while putting a first call on hold. After obtaining the second dial tone, the basic procedure format is followed. The KRO signal for a three-way connection is received as a request to switch the talk channel.

The parameters of additional information signals are presented in Table 1.

The basic volume of DVO system data is for subscriber characteristics (AKh) or the translator for decoding the line number of the subscriber (LNA) to a set of finite parameters such as the listed number of the subscriber, the type of subscriber line (AL) and its relation to the different forms of equipment. The basic structural principle of the translatoris organization of multistep indexed tables, as a rule, of no more than three steps.

The first translator stage is aone-dimensional matrix of so-called first words of the AKh. The size of the matrix is uniquely determined by the finite subscriber capacity of the office and fully corresponds to it. The initial parameter for indexing the given table is the complete LNA in binary code. The unincluded AL are denoted by "all 1's." The entries corresponding to included AL contain the type P descriptor (the high-order bit of the word) which indicates the nature of the information in the rest of the word. For P=0 the remainder contains data; for P=1 the remainder contains a reference to the next step of the AKh. The second and following steps of the AKh can have altered format.

For realization of some services, for example, "reminder" and "return call," connected single-pass lists of variable length are used which are formed of four-word blocks in the ready-access memory. In addition, from the second step of the AKh of the service user there is a reference to this block or several blocks and one of the lists. In all, there are four types of such lists: the free block list, the list scanned once every second, the list scanned once every minute, the list for organizing the printing out of the information from the DVO system.

The DVO system programs participate in the following calling phases: reception of the number from the subscriber, arrival of a call at the subscriber, arrival of the disconnect signal from the subscriber and organization of sending special signals to the subscriber. In addition, there are a number of programs, the so-called operation planners triggered by a timer once a second, minute and hour on the basic program level. The DVO software, just as the rest of the

Table 1

Position No	Name of signal	Frequency, hertz	Time parameters: Sending time	Time parameters: Pause time
1	Special "office answer" signal	425±25	0.3±0.1	Single sending
2	Confirmation of reception of request	425±25	Continuous signal	
3	Confirmation of implementation of a request from a call	425±25	1.5±0.5	With subsequent return to the call
4	Failure to receive the request	425+25	0.3–0.4	0.3–0.4
5	Failure to implement the service from the call	425±25	0.3–0.4 For 1-2 seconds with subsequent return to the call	0.3–0.4
6	Notification of a new incoming call during conversation	425+25	0.2±0.1	5±1
7	Making a call with "return call" service	25±2	1±0.25	1±0.25
8	Making a call with "reminder" service	25±2	2±0.5	2±0.5
9	Signal of an incoming "urgent" call	425±25	0.15±0.05 three sendings	0.15±0.05

ATS KE software, is constructed on the modular principle in the form of a common program. This means that the programs are written without being tied to the physical addresses of memory and can be moved. The references between the programs and between the programs and the office data are by means of reference tables. Thus, when generating the ATS KE software only the office data, for example, the AL repeaters, SL repeaters, the routings, and so on describing the specific office and coupling it to the communications network are subject to generation.

The memory used to implement the DVO is presented in Table 2.

Table 2

Name of set of operations	Volume (K-words)
Design operations	1.0
Register information processing	3.1
Implementation of DVO in the incoming call phase to a subscriber and organization of special signal sending	1.5
Processing of disconnect signals from the subscriber and KRO	3.0
AKh for using the DVO for 2048 AL capacity	16.0
Total:	24.6

Introduction of the developed DVO system has the following advantages: it expands the subscriber capabilities; it insures economy of personal and work time; the possibility of transition if desired from a system with rejects to a system with waiting; increases the carrying capacity of the ATS as a result of partial elimination of repeated calls, and it increases the carrying capacity of the recording equipment.

CHOICE OF DYNAMIC CONTROL METHOD FOR INFORMATION FLOWS ON A COMMUNICATIONS NETWORK

[Article by Yu. M. Kazachenko, Leningrad, pp 71-75]

One of the ways to increase the operating efficiency of a communications network is to optimize the control of the interaction of individual network subsystems and the control subsystem itself. Therefore recently significant attention has been given to investigation of adaptive control of information flow distribution in the communications networks. The realization of this control faces the designers with at least the two following problems:

The placement of the control subsystem (centralized, decentralized);

Organization of service information exchange, the rate of renewal of service information or the adaptation rate of the control algorithm if the service information exchange is not carried out.

Theoretical analysis [1] and also the results of experiments on existing networks [2, 3] permit isolation of three basic situtions which arise during the operation of a communications network: small loading, loading close to critical and overloading. In each situation the efficiency of the information traffic control will be different. As is noted in [1, 2, 3], the greatest effect is obtained from the application of adaptive information traffic control (dynamic routing) in the presence of significant loads. With a small load, static routing is effective; for overloads restricting the load coming from the network subscribers appears to be the only possible solution (without altering the structure of the network). Thus, quasistatic control will be optimal [4].

In this case the service information exchange rate will be relatively low [3], which will lower the nonproductive load on the channels. Let us note that in the ARPA network with fully decentralized, dynamic control up to 50% of the carrying capacity of the communications channels is used to transmit service data [5].

The centralized and decentralized methods of constructing the control subsystem are characterized by advantages and disadvantages [4]. Let us only note that during overloads (that is, restriction of the incoming flows) the control is the most effective for the centralized method [6]. Thus, an intermediate method of placement of the information traffic control subsystem is expedient [7, 8]. One version of the structure of a control subsystem can be as follows.

Let us consider a network of N junctions with arbitrary connections. Let $Q_{i,j}(t)$ be the quantity of information at the time t which must be transmitted from junction i to junction j. The values of $Q_{i,j}(t)$ form an N×N matrix Q(t) describing the "operation" which must be carried out in the network at the time t. The variation rate of the matrix Q(t) is determined by the variation rate of the total quantity of information in the network. Also let $c_{i,j}(t)$ be the carrying capacities of the channel directly connecting junctions i and j; C(t) is the matrix describing the reserves which are available in the network for performance of the operation Q(t). Let us propose that C(t) is a two-dimensional file of elements $c_{i,j}(t)$. Now the routing problem consists in optimal distribution of reserves C(t) for execution of the operation Q(t). In this case the current delay will be

$$d^*_{i,j}(t)=Q_{i,j}(t)/c_{i,j}(t). \tag{1}$$

The average delay is calculated in the switching center

$$d_{i,j}(t)=k\cdot d_{i,j}(t-\tau_\alpha)+(1-k)d^*_{i,j}(t), \tag{2}$$

where $0\leq k\leq 1$ and τ_α characterizes the history.

The value of $d_{i,j}(t)$ is transmitted to the network control center (TsUS) only if

$$|d_{i,j}(t)-d_{i,j}(t-l\,\tau_\alpha)|>d^{(1)}_{\text{пор}}, \tag{3}$$

Key: 1. threshold

where L=1,2,3,... which corresponds to quasistatic control.

The general state of the network is estimated at the TsUS on the basis of the matrix D(t) with the elements $d_{i,j}(t)$. The shortest path is found using a method, for example, the dynamic programming method. Here not one, but n shortest paths are defined. Let $\Pi^k_{i,j}$ denote the k-th path of the investigated paths joining the junctions i and j. For this path the delay will be

$$T^k_{i,j}(t)=\sum_{[q,z]\in\Pi^k_{i,j}} d_{q,z}(t). \tag{4}$$

The value of (4) is calculated for all n routing versions and the values of $T^k_{i,j}(t)$ are compared to each other. If one value is distinguished from another by less than some amount δ, control is realized by the switching center control subsystem. For example, the method of dynamic priorities car be used [9], where the information distribution plan on the network does not change. In the opposite case the TsUS selects the shortest path and corrects the information distribution plan. The nature of the solutions adopted when using the given method can be demonstrated in an example where the number of paths between the i and j junctions from which it is possible to choose is n=2. At the TsUS, the delays are compared with respect to these paths and

$$\text{if } T^{k=A}_{i,j}(t)<T^{k=B}_{i,j}(t)-\delta\quad,\ \text{then } L_{i,j}(t)=A\ ; \tag{5}$$

$$\text{if} \quad T^{k=B}_{i,j}(t) < T^{k=A}_{i,j}(t) - \delta \quad \text{then} \quad L_{i,j}(t) = B; \tag{6}$$

$$\text{if} \quad |T^{k=A}_{i,j}(t) - T^{k=A}_{i,j}(t)| \leqslant \delta, \text{ then} \quad L_{i,j}(t) = A, B. \tag{7}$$

Thus, in the cases (5) and (6) the routing is selected on the TsUS, and in the case (7), the path is chosen at the switching center on the basis of the routing matrix developed in the network design stage. The parameter δ determines the degree of independence of the switching junction. If δ is small (that is, even insignificant changes are considered in the delays with respect to different paths), we arrive at a fully centralized control. If δ is large, we arrive at decentralized path selection. Exceeding some threshold by the average delay with respect to the entire network indicates overloading of the network with messages. In this case the TsUS goes to the load limiting algorithm, for example, curtailing access of low-priority messages to the network or using other load limiting algorithms [4].

Thus, implementation of the discussed method permits the use of the advantages of decentralized and centralized methods of constructing the control subsystem. In addition, the given method permits efficient use of the control subsystem in the entire range of load variation.

One of the primary difficulties in implementing any nontrivial method of dynamic traffic control on a communications network is the fact that the "operation" $Q(t)$ is distributed in space and time. In other words, the special operation $Q_{i,j}(t)$ is known exactly only at the time t and only at the junction i. Information about $Q_{i,j}(t)$ can be transmitted to another junction, but this information can become obsolete during transmission time. Thus, any global characterization of the general operation of the network $Q(t)$ is based on data pertaining to the past and not current information about the state of the junctions. In the network there are two types of information about its state:

Precise and timely local information (that is, information about the state of the switching center at the same center). This information is available for every center;

Average values describing the past local characteristics of the network.

The proposed dynamic control method permits full use of both types of information about the state of the network, which insures effectiveness of it.

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SELECTING TYPES OF PROCESSORS FOR A MULTIPROCESSOR SYSTEM

[Article by A. N. Kol'tsov and F. I. Pepinov, Moscow, pp 75-77]

A study is made of a multilevel hierarchical system with given structure. The apexes of different level joined to each other form a precursor -follower pair. In the k-level system the i-th apex of the ℓ-th level forms the set F_i of its followers of the $\ell+1$ level and the set F_i^* of its followers from $\ell+1$ to k level. This system can be a model for execution of a parallel computation problem, administrative relations and the relations of any agency, a multiprocessor data processing system, control unit, and so on [1]. Beginning with the selected structure of the system and its purpose, a special algorithm σ_i is matched to each apex of the system i in some way or another. This special algorithm is executed by some type of processor from among the given ones located at the i-th apex of the system. The composition of the special algorithms $\sigma=\{\sigma_1,\ldots,\sigma_m\}$ is the operating algorithm of the system.

In the investigated system in the processor memory of the i-th apex, in addition to the special algorithm σ_i executed on this processor, all special algorithms σ^*_i matched to the follower apexes from the set F^*_i are stored (but not executed). The special algorithm stored in the memory of the i-th apex processor can be transmitted to the follower apexes from the set F_i.

The investigated system is implemented in a set of processors of different types $R=\{R_1,\ldots, R_n\}$. The types of processors R_j, $j=1,\ldots,n$ can differ significantly with respect to their capabilities; therefore each special algorithm σ_i, $i=1,\ldots,m$ can be executed on one type of processor or another with different speed, reliability and cost. The execution of σ_i on R_j is characterized by the following parameters of the special algorithms and processors.

C_j is the cost of the j-th type of processor; λ_j is the intensity of failures of the j-th type of processor; t_i is the given execution time of σ_i; L_i is the magnitude of the penalty for failure to execute the algorithm σ_i; V_i is the magnitude of the penalty for exceeding the execution time of the algorithm σ_i per unit time; p_{ij} is the probability of failure to execute the algorithm σ_i when executing it on the j-th type processor; c_{ij} is the cost of developing a method of implementing the special algorithm σ_i on the j-th type processor; t_{ij} is the time of execution of σ_i on the j-th type of processor.

The enumerated parameters will permit determination of the magnitude of the average losses ψ_{ij} when executing the algorithm σ_i on the j-th type processor which is made up of the following components:

c'_{ij} is the cost of executing the algorithm σ_i on the j-th selected type processor, $c'_{ij}=c_j+c_{ij}$;

$L_i p_{ij}$ is the average magnitude of the penalty for failure to execute σ_i, where the probability of this event is calculated by the formula:

$$p_{ij} = 1-e^{-\lambda_i t_{ij}};$$

$V_i t'_{ij}$ is the penalty for exceeding the execution time of the special algorithm σ_i by the time t'_{ij} which is calculated as follows:

$$t'_{ij} = \begin{cases} 0, \text{ if } & t_{ij} \leq t_i \\ t_{ij}-t_i & , \text{ if } \quad t_{ij}>t_i. \end{cases}$$

The magnitude of the average losses ψ_{ij} when executing the special algorithm σ_i on the j-th type processor is calculated by the formula:

$$\psi_{ij} = L_i \rho_{ij} + V_i \cdot t'_{ij} + C'_{ij}. \tag{1}$$

The magnitude of the average losses in implementing the entire system:

$$\psi = \sum_{i=1}^{m} \psi_{ij} \tag{2}$$

It is obvious that the magnitude of losses ψ depends on which type of processor is used to execute each of the special algorithms $\sigma_i \in \sigma$.

A method of selecting the types of processors executing the set of special algorithms is proposed such that the magnitude of the average losses ψ for the given hierarchical multilevel system will be minimal. Here it is remember that the number of processors in the system is no less than the number of special algorithms

$$m \leq \sum_j r_j, \tag{3}$$

where r_j, $j=1,\ldots,n$ is an arbitrary given number of processors of the j-th type. Otherwise, it is also possible to combine some special algorithms so that condition ((3) is met.

For each special algorithm $\sigma_i \in \sigma$, values of the average losses $\psi_{i1}, \psi_{i2}, \ldots, \psi_{in}$ are calculated by formula (1) for execution of the special algorithm on all given types of processors $R_1, R_2, \ldots, R_n$, that is, each algorithm σ_i has a row of losses $\psi_{i1}, \psi_{i2}, \ldots, \psi_{in}$ in correspondence to it. The set of such rows forms a loss matrix ?, each column of which is assigned an r_j — the number of type R_j processors.

On the basis of the loss matrix Ψ, a matrix breaking down the special algorithms with respect to types of processors B is constructed, the rows and columns of which precisely correspond to the loss matrix, and the elements are b_{ij}, where

$$b_{ij} = \begin{cases} 0, \text{ if } \sigma_i \text{ is not executed on the } R_i \text{ processor} \\ 1, \text{ if } \sigma_i \text{ is executed on the } R_j \text{ processor.} \end{cases}$$

Inasmuch as each special algorithm is executed on only one processor,

$$\sum_{j=1}^{n} b_{ij} = 1. \tag{4}$$

On the other hand, the number of processors of each type R_j is given, and this means that

$$\sum_{i} b_{ij} \leqslant z_j. \tag{5}$$

The expected magnitude of the losses Ψ for the entire system with respect to all special algorithms is defined as follows:

$$\Psi = \sum_{j} \sum_{i} (b_{ij} \cdot \Psi_{ij}). \tag{6}$$

The obtained expression (6) is the solution functional of the system, the minimum value of which determines the choice of types of processors for the entire system with the existing restrictions (4) and (5).

HOMEOSTATIC PRINCIPLE OF REGULATING THE OUTGOING SUBSCRIBER TRAFFIC

[Article by A. V. Kotov, Leningrad, pp 77-82]

Within the framework of dynamic traffic control on telephone networks, the volume of the outgoing traffic from the subscribers can be automatically regulated. This is extremely desirable from the point of view of increasing the cost effectiveness of the communications networks [1, 2]. However, implementation of the indicated regulation is possible only under the condition of improving the algorithm for interaction between calling subscribers and the telephone system in which negative feedback is introduced into the "subscriber-telephone network" control circuit. The conversion of the interaction algorithm with introduction of negative feedback into the telephone system is realized, for example, by certain additional types of services such as the automatic answering device, "putting the call on hold" and so on. These services decrease the number of repeated calls (PV) and thus have an automatic regulating effect on the outgoing traffic from the subscribers. Inasmuch as the largest number of PV on a network are formed as a result of low accessibility of such GTS services as ticket ordering, the information office, and so on, it must be expected that the greatest regulating effect with respect to the occurring load is provided by services which are designed to eliminate rejects on the part of the indicated services. An example of a hardware solution in this area is described in [3]. Here the subscriber calling for service gets a recorded message about the time by which he will receive a return call for servicing his request. The indicated time depends on the number of requests accumulated previously for servicing.

As is known, in ordinary request and information service systems the load is characterized not only by the fact that it has a sharply fluctuating intensity, but also by the fact that even during short-time intervals, for example, in the plh [peak load hour] it does not have the properties of a simplest flow in view of the presence in it of a large number of PV (a flow with consequence). In contrast to this, in the system according to [3], the total load is somehow split by the regulating mechanism into two flows: 1) a simplest flow (flow without PV) with low intensity $y_{occur} \ll y_{max}$ consisting of the actual service requests coming into the system in real time and having a constant service time (transmission of a voice message) $\tau \ll t_{ave}$ (where t_{ave} is the average busy time in case of ordinary requests for information services), 2) an ordered (deterministic) flow of request servicing with a time shift having an intensity y=const.

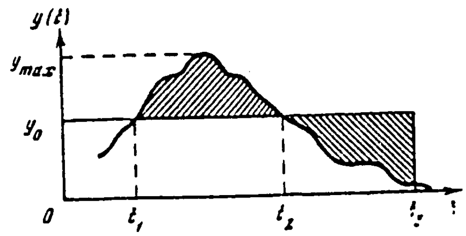

Figure 1

In order to explain the essence of the given regulation principle let us consider Figure 1 which depicts the load intensity distribution curve $y=y(t)$ of the request or information service load for some time interval. The level y_0 denotes the total carrying capacity of the entire operator staff (depending on the time of day this level can vary in accordance with some schedule). If the service is equipped with a system according to [3], after a time t_1 all of the calls begin to be serviced with waiting. Up to the time t_2 (that is, while $y>y_0$) the queue is enlarged and after t_2, it diminishes. The end of servicing with waiting takes place at some time t_3 defined from the expression

$$\int_{t_1}^{t_2}[y(t)-y_0]\,dt=\int_{t_2}^{t_3}[y_0-y(t)]\,dt\,.$$

In Figure 1 this corresponds to equality of the crosshatched sections of the area above and below the curve $y=y(t)$. A call coming into the system at the time t_2 will be serviced with maximum waiting time T_{max} equal to

$$T_{max}=\frac{1}{y_0}\int_{t_1}^{t_2}[y(t)-y_0]\,dt\,. \qquad (1)$$

Expression (1) is a mathematical model of a telephone system with self-regulation with respect to subscriber traffic directed to information or request services. The regulation effect in the given case consists in the fact that for given carrying capacity of the system y_0, whatever the intensity of the arrival of requests (of course, within known limits), the service waiting time T_{max} will always be automatically selected, in which the intensity of the traffic service by the operators remains constant and equal to y_0. Here the total traffic intensity $y_{occur}+y_0$ will be appreciably less than y_{max}; all of the calls will be serviced, and no rejects or PV will occur.

Another possibility for controlling the outgoing traffic from subscribers consists in using the mechanism of adaptive rates on the networks [2]. The essence of this method consists in the fact that a system of variable rates is introduced on the telephone network, the size of which is determined by the load intensity of the ATS, and negative feedback is realized in the form of operative transmission of information about current rates to the subscribers. In this case a regulating effect on the occurring load as a whole is realized. The given method of regulation is based on sifting out part of the load (transferring it from peak periods to other times) by the criterion of quantitative consideration of the human factor -- the requirement for communications under various specific conditions. This factor can be quantitatively expressed in terms of the value content of the information in the calls.

The random variable θ -- the value content of information in a call -- is defined as a subjective estimate by the calling subscriber of the value (expressed in money) of this information which the subscriber hopes to transmit, receive or exchange as a result of making a given specific call. Figure 2 shows the hypothetical probability density distribution curve for the indicated variable.

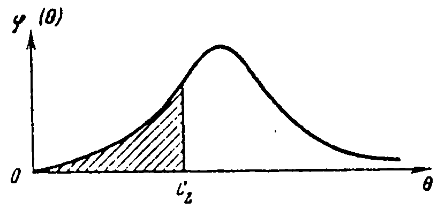

Figure 2

The mechanism of adaptive rates presupposes that if the measured intensity of the occurring load does not exceed a threshold value y_π, the preferential rate c_1 is in effect, for example, equal to zero, and the subscribers are informed of this by the corresponding signal. When an increase in traffic intensity exceeds y_π, an increased rate c_2 is automatically put into effect on the network, and the signal transmitted to the subscribers changes simultaneously. With a further increase in traffic, rate c_3 can be put into effect and the signal corresponding to it, and so on. If, for example, rate c_2 has been put into effect, the calls for which the value content of the information $\theta<c_2$, that is, the calls corresponding to the crosshatched part of the area under the curve $\phi(\theta)$ (see Figure 2) will not be made by the subscribers, and they will be postponed to a later time when the rate c_1 will again be in effect as a result of a drop in the load on the network. This part of the calls will be:

$$\delta_1 = \int_0^{c_2} \varphi(\theta)\, d\theta .$$

The rest of the calls are made. Inasmuch as the entire area under the curve $\phi(\theta)$ is equal to one, the proportion of calls made by the subscribers during the period that the rate c_2 is in effect will be:

$$\delta_2 = 1 - \int_0^{c_2} \varphi(\theta)\, d\theta .$$

The load intensity proportional to the entire area under the curve $\phi(\theta)$ characterizes the total demand for the entire set of subscribers for communications at the investigated point in time. Let us denote this value by y_{demand}. Then the load intensity occurring at the same point in time in the presence of the adaptive rate mechanism will be expressed as follows:

$$y_{\text{возн}}^{(1)} = y_{\text{потр}}^{(2)} \left(1 - \int_0^{c} \varphi(\theta)\, d\theta\right), \qquad (2)$$

Key: 1. occur; 2. demand

where c is the current value of the rate.

Expression (2) is the regulation function -- a model of the telephone network equipped with an adaptive rate mechanism. The regulation effect consists only in the fact that whatever the peak value of the demand of the entire set of subscribers for communications it is always possible to select a rate c_i for which the occurring load will be less than some previously given value $y_{occur\ \pi}$.

The form of the function $\phi(\theta)$ is still unknown at the present time. However, there are grounds for assuming that in the given case normal distribution occurs. Under this assumption the network model assumes the form:

$$y^{(1)}_{\text{возн}} = y^{(2)}_{\text{потр}}\left(1 - \frac{1}{\sigma\sqrt{2\pi}}\int_0^c e^{-\frac{(\theta-\theta_m)^2}{2\sigma^2}}\,d\theta\right),$$

Key: 1. occur; 2. demand

where θ_m is the mathematical expectation of the value of θ, σ is the mean square deviation.

Both of the investigated methods of regulating the outgoing traffic from the subscribers are based on applying technical solutions that introduce negative feedback into the man-machine "subscriber-telephone network" system. As is known [4] the negative feedback is a necessary condition for giving a complex probability system the properties of homeostasis, that is, the capacity to withstand dynamic (fluctuating within some sufficiently narrow limits) constancy of its significant variable under the conditions of sharp inconstancy of external disturbances affecting the system. By significant variable we mean the variable closely connected with the operating quality of the system. In our case it is the occurring telephone traffic which, as is known, in the absence of regulating mechanisms is distinguished by extraordinary fluctuation with respect to hours of the day, days of the week, and so on. Complex probability systems having the homeostasis mechanism belong to the class of cybernetic systems [4]. Therefore the application of the above-described technical solutions on telephone networks and also the DVO which introduce some degree of negative feedback into the "subscriber-telephone network" system gives the latter the properties of a cybernetic system operating in the optimal version. All of the mentioned technical solutions can be used both separately and simultaneously. In the latter case it is necessary to expect that a maximum overall self-regulating effect will be obtained.

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RESEARCH OF THE INTERNATIONAL TELEPHONE AND TELEGRAPH CONSULTATIVE COMMITTEE IN THE FIELD OF NETWORK CONTROL

[Article by A. V. Kotov, Leningrad, pp 82-85]

Automation of international telephone communications, growth of information exchange are raising the urgent problem of introducing methods of controlling the traffic flows on international networks which previously, in the absence of automation, were realized by international telephone operators. However, the problem of organizing work control on an international scale not only has technical and economic aspects, but it also requires the organization of cooperation of the communications administrations of the participating countries. It is natural that the solution of this type of problem is only possible for an international communications administration organization -- the International Electrocommunications Union (MSE). One of the permanent agencies of the MSE is the International Telephone and Telegraph Consultative Committee (MKKTT) which at the present time (the investigated period is 1977 to 1980) includes 17 investigative commissions and more than 10 mixed, regional and other specialized working groups.

Research in the field of traffic flow control in international networks has been performed by the MKKTT for a number of years. In the preceding research period (1973 to 1976) 13 research committees were engaged in this research. The result of earlier research was the Recommendations Q.55/E.410 approved by the Fourth Plenary Meeting of the International Telephone and Telegraph Consultative Committee in 1972 [1]. This recommendation substantiates the necessity for controlling traffic flows on international networks, the concept of work control is defined, and the actions required for implementation of it (performed manually or automatically) are enumerated.

The purposes of network control are indicated as insuring continuity of communications and maximizing the number of paid calls as a result of the most complete possible use of equipment both during normal operating periods of the network and during periods when there is some disturbance in its operation.

The Recommendation includes a description also of the parameters (indices, observation of which will permit determination of the time when it is necessary to take measures to control the network and establish what actions it is expedient to take. These parameters are as follows:

An overload coefficient (expressed in percentages) defined as the ratio of the number of unsatisfied requests for a free channel in any group of channels (circuits) to the total number of requests coming to this group during a defined time interval;

The BCH index defined as the ratio of the average number of requests for free channels coming to a given group of channels during 1 hour to the number of channels in the group;

The SCH index indicating the average number of busies during the hour per operating channel of the given group.

An important item of the Recommendation Q.55/E.410 is indication of the necessity for developing a system of signals which must be transmitted from certain switching centers to others during the network control process.

In the same Recommendation it is stated that the time for organizing any international center in which the control of the international network would be concentrated still has not come and that on the modern level, the organization of load control in international service must be realized on the basis of two-way or multisided mutual agreements between the communications administrations of various countries.

During the next research period (1973 to 1976) the problem of load traffic control on the international networks became the content of Question 4/XIII. A very important new measure of this research period was the organization of an international experiment in introducing load flow control under actual conditions -- in existing international service. Among the purpose of the experiment were the following: 1) obtaining experience in using network control on the international level; 2) determination of the threshold values of the indices used in network control; 3) development of proposals regarding new indices which would be useful in the sense of increasing the control efficiency; 4) the development of proposals with respect to further improvement of the system of signals transmitted between the switching centers during the load control process, and so on.

For countries agreeing to participate in the experiment, the form of periodic reports on the course of the experiment was developed. The report data were generalized and published in the documents of the MKKTT by the person reporting on Question 4/XIII. The results of the experiment which was participated in by several countries were generalized and presented for investigation by the Sixth Plenary Meeting of the MKKTT which was held in 1976. They were published in the first official document of the MKKTT in the current research period 2. Recommendation Q.55/E.410 was approved and put in Volume II.2 of the MKKTT Orange Book. The decision was made to continue the international experiment in a new research period (1977 to 1980).

The Sixth Plenary Meeting made changes in the composition of the research committees. The 13th Committee was formed, and the greater part of its problems, including the problem of network control, were transferred to the II Committee.

During the current research period network control is the content of Problem 18/II [2]. The primary work is concentrated around further continuation and expansion of the international experiment. By December 1977, the number of countries participating in the experiment reached 12.

At the present time the Recommendation Q.55/E410 is being reexamined. A new altered, supplemented version of it containing an appendix with definition of terms has been published in [3]. It was presented for approval by the Seventh Plenary Meeting of the MKKTT in 1980.

The MKKTT is making an effort to involve new countries in the participation in the experiment with respect to network control on an international level and is paying special attention to the necessity for more precise definition of the threshold values of the indices which were obtained during the course of previously performed work and also the necessity for developing an optimal signal system considering both the requirements on the part of network control and other requirements on the signalling system, the development of which enters into the competence of the Eleventh Committee of the MKKTT. The following signals have been proposed.

Information signals: 1) the switching center is overloaded as a result of overloading common devices; 2) the SN index exceeded the threshold value in a given group of networks; 3) blocking of the lines or switching centers of the national network occurs on the routings (the codes are indicated); 4) the location of the blocking of the lines or switching centers of the international network occurs on the routings (codes are indicated).

The signals transmitting the information and requiring that special measures be taken with respect to network control are as follows: 1) blocking has occurred in the switching center. It is proposed that the load be reduced by 25, 50, 75 or 100% (one of these figures is indicated); 2) blockings in the line groups or in the switching centers of the national network have arisen in the stated routing (the codes are indicated). It is proposed that the load be reduced by 25, 50, 75 or 100% (one of these figures is indicated); 3) the same as 2), but for the international network; 4) the signal (indicated by the signal identifier code) is cancelled; 5) the proposed action is taken.

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SOME CHARACTERISTICS OF PACKET-SWITCHED DATA TRANSMISSION SYSTEMS

[Article by V. N. Koshelev, Moscow, pp 85-90]

The purpose of this paper is to investigate the dependence of the mathematical expectation of the time spent by a message broken down into packets in a queue, for the packet switching method (KP) with priority parameter assigned to it by p -- from the length of the packets, the number of packets and the number of outgoing channels from the UK. At the same time, a study will be made of the message parameters which determine the choice of the length of the service-address information (SAI), and the load was analyzed. The results of the analysis can be used to select the number of packets into which it is necessary to break down the message for a given number of outgoing channels from the UK.

Let us consider the dependence of the increase in message length for KP on the increase in SAI volume. Let us assume that the length of the title of the packets is approximately equal to the length of the titles of the messages when switching messages (KS); then:

$$\frac{1}{\mu_c} = m \cdot \frac{1}{\mu_n} - (m-1) H_з , \tag{1}$$

where $1/\mu_c$ is the average message length with a title;

$1/\mu_\pi$ is the average packet length with a title;

H_3 is the title length;

m is the number of packets in the message.

Let us denote the α-coefficient of the length of the packet title indicating how much the title length is less than the packet length:

$$\alpha = \frac{1}{\mu_n H_з} , \tag{2}$$

then

$$H_з = \frac{1}{\mu_n \cdot \alpha} . \tag{3}$$

From (1) and (3), it is possible to obtain the expression relating the average length of the complete message and the message broken down into packets:

$$\frac{1}{\mu_c} = \frac{m(\alpha-1)+1}{\alpha\mu_n} \qquad (4)$$

or

$$\frac{1}{\mu_c} = \beta\frac{1}{\mu_n}, \qquad (5)$$

where

$$\beta = \frac{m(\alpha-1)+1}{\alpha}; \qquad (6)$$

β is the packeting coefficient indicating how long the message broken down into packets is with KP greater than the length of the same message with the KS method.

Analysis of the coefficients α and β for different values of $1/\mu_\pi$, H_3, m indicates that the message informativeness coefficient for KP:

$$i = \frac{\beta}{m} = \frac{m(\alpha-1)+1}{m\alpha} \qquad (7)$$

depends on α so that $i \to 1$ for $\alpha \to \infty$. The graph of this function is presented in Figure 1.

The results of the analysis of α and i can be used when selecting H_3 of the packet.

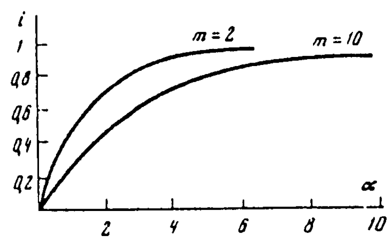

Figure 1

The KP method permits transmission of the packets of one message simultaneously over different channels. This leads to the presence of two cases influencing the message servicing time for KP:

The service time is equal to the packet service time ($N \geq m$);

The message servicing time is longer than the packet servicing time, but smaller than the message servicing time for KS ($N<m$). It is obvious that in the general case the packets of one message will be serviced on the UK in several cycles equal to the following:

$$z = W\left|\frac{m}{N}\right|, \qquad (8)$$

where W is the integer from dividing m/N with the remainder,

N is the number of outgoing channels from UK.

The intensity of the arrival of messages broken down into packets for servicing for KP will be: $\lambda_{c\pi}=r\lambda_{\pi}$, and the load on the system when processing message is broken down into packets:

$$\rho_{cn}=r\rho_{n}, \tag{10}$$

where ρ_n is the load on the system when transmitting packets in one operating cycle of the system.

$$\rho_n=\frac{\rho_c}{\beta}. \tag{11}$$

The ratio of the loads on the system when using the KP on KS methods will be:

$$\rho_{cn}=\frac{r\rho_c}{\beta}. \tag{12}$$

Analysis of the interrelation of $\rho_{c\pi}$ and ρ_c for different m and N is illustrated in Figure 2, and it leads to the conclusion that increasing the number of packets (m) in a message leads to reduction of the $\rho_{c\pi}$ with constant N. Increasing the number of packets in the message which increases the number of service cycles (r) leads to a reduction in the upper bound of $\rho_{c\pi}$. Increasing the number of outgoing channels (N) from UK offers the possibility of lowering the lower bound $\rho_{c\pi}$. The results of analyzing $\rho_{c\pi}$ can be used to select the number of packets in a message for the given number of outgoing channels from the UK.

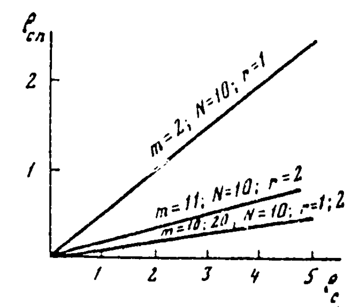

Figure 2

Let us consider the variation of the average waiting times in the different priority systems when using the KP method ($W^s_{c\pi p}$), and let us compare them with the mean waiting times for KS (W^s_p). Here s is the priority system index. The mean waiting time graphs were constructed for the KS by the control systems for them taken from [1].

The average message length broken down into packets will be:

$$\frac{1}{\mu_{cn}}=\frac{m}{\beta\cdot\mu_c}. \tag{13}$$

For a system with fixed priority without interrupt it is possible to write:

$$W^{\varphi}_{cnp} = \begin{cases} \dfrac{\int_{cn} \cdot \rho_{cn(j-1)}/\mu_{cn(j-1)} + \sum_{i=j}^{p} \rho_{cni}/\mu_{cni}}{(1-\sum_{i=p+1}^{p} \rho_{cni})(1-\sum_{i=p}^{p} \rho_{cni})} & \text{for } p \geq j \\ \infty & \text{for } p < j, \end{cases} \tag{14}$$

where j is a minimum positive integer such that

$$\sum_{i=j}^{p} \rho_{cni} < 1, \tag{15}$$

$$\int_{cn} = \begin{cases} 0 & , \rho_{cn} < 1 \\ \dfrac{1-\sum_{i=j}^{p} \rho_{cni}}{\rho_{cn(j-1)}} & , \rho_{cn} \geq 1, \end{cases} \tag{16}$$

p is the number of different classes of priorities.

The family of characteristics μW^{ϕ}_{p} and $\mu_{c\pi} W_{c\pi p}$ is depicted in Figure 3.

For a system with fixed priority with interrupt for $\rho_{c\pi} \geq 0$, we obtain:

$$W^{\varphi n}_{cnp} = \begin{cases} \dfrac{\frac{\rho_{cn}}{\mu_{cn}} + \sum_{i=p+1}^{p} \rho_{cni}\left(\frac{1}{\mu_{cnp}} + \frac{1}{\mu_{cni}}\right) + \sum_{i=p+1}^{p} \rho_{cni} W_{cni}}{1-\sum_{i=p}^{p} \rho_{cni}} & , p \geq j \\ \infty & , p < j. \end{cases} \tag{17}$$

The family of characteristics $\mu W^{\phi\pi}_{p}$ and $\mu_{c\pi} W^{\phi\pi}_{c\pi p}$ is depicted in Figure 4.

For systems of priorities with dependent delay, the mathematical waiting time which is required for completion of servicing of the message broken down into packets which is in processing, on entry of the new message into the system of packets considering (10) and (13) will be:

$$W_{cno} = \frac{2m}{\beta^2} \sum_{i=1}^{p} \frac{\rho_p}{\mu_p}, \tag{18}$$

where $W_0 = \sum_{i=1}^{p} \rho_p/\mu_p$ is the mathematical expectation of the time which is required for completion of servicing of a complete message in processing, on entry of a new message into the system [1]. The growth rate of the priorities b_p for the system of priorities with dependent delay [1] for KP remains the same as for KS.

For the system of priorities with dependent delay without interrupt for $0 \leq \rho_{c\pi} < r/\beta$

$$W^{з}_{cnp} = \frac{W_{cno}/(1-\rho_{cn}) - \sum_{i=1}^{p-1} \rho_{cni} W_{cni} (1-b_i/b_p)}{1-\sum_{i=p+1}^{p} \rho_{cni}(1-b_p/b_i)}. \tag{19}$$

The family of characteristics μW^{3}_{p} and $\mu_{c\pi} W^{3}_{c\pi p}$ is depicted in Figure 5.

For a system of priorities with dependent delay with interrupt for $0 \leq \rho_{c\pi} < r/\beta$

$$W^{зп}_{спр} = \frac{1}{1-\sum\limits_{i=p+1} \varrho_{спi}(1-b_p/b_i)}\left[\frac{W_{спо}}{1-\varrho_{сп}} + \sum_{i=p+1}^{p} \frac{\varrho_{спi}}{\mu_{спi}}\left(1-\frac{b_p}{b_i}\right) - \sum_{i=1}^{p-1} \frac{\varrho_{спi}}{\mu_{спi}}\left(1-\frac{b_i}{b_p}\right) - \sum_{i=1}^{p-1} \varrho_{спi} W_{спi}\left(1-\frac{b_i}{b_p}\right)\right] . \quad (20)$$

The family of characteristics $\mu W^{3\pi}_{c\pi p}$ and $\mu_{c\pi} W^{3\pi}_{c\pi p}$ is depicted in Figure 6.

Analysis of the graphs in Figures 3-6 indicates that the mathematical expectation of the time spent by the messages broken down into packets in the queue for KP is appreciably less than the mathematical expectation of the time spent by the messages in the queue for KS. It must be noted that for all the priority systems the messages broken down into packets have a finite value of $W_{c\pi p}$ for all categories of priorities for saturation of the servicing device ($\rho=1$). The application of KP for the systems of priorities with dependent delay turns out to be especially useful, for with KS for all groups of priorities they have $W_p \to \infty$ for $\rho_c=1$, and for KP, for all the priority groups they have finite, insignificant times spent by the messages in the queue.

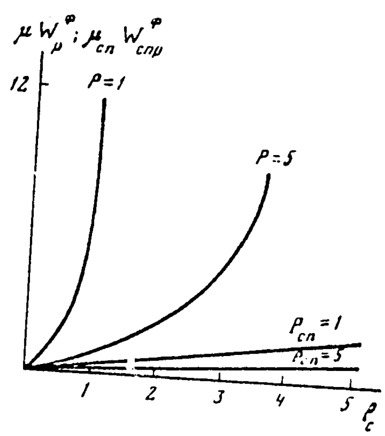

Figure 3

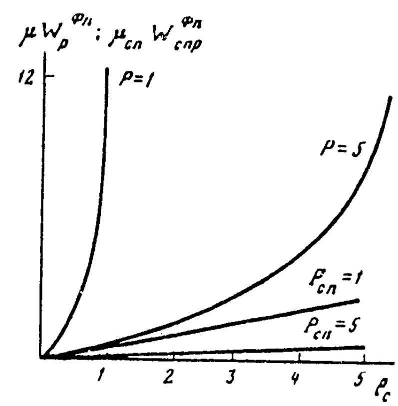

Figure 4

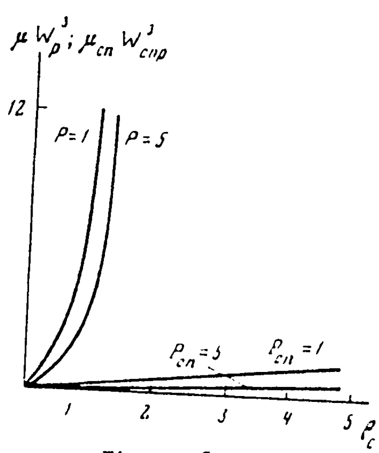

Figure 5

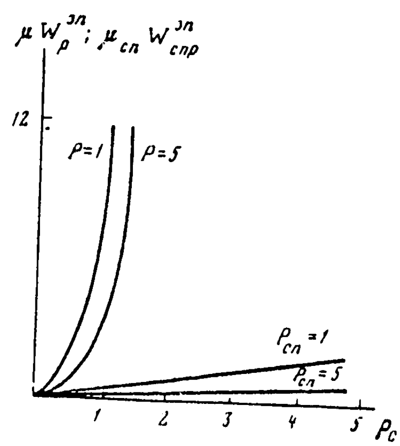

Figure 6

Analyzing the graphs in Figures 3-6 jointly with Figure 2 it is possible to draw the conclusion that increasing N, m, β leads to a decrease in $W_{\text{спр}}$.

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SIMULATION OF PROGRAMMED CONTROL PROCESSES IN SWITCHING CENTERS

[Article by Ye. V. Konovalov, Minsk, pp 90-93]

The purpose of the studies of the control processes in the switching centers (KU) is development of recommendations, methods and instruments for the KU developers. However, active use of the results of these studies in design practice is being held up for a number of reasons, among which inadequateness of the investigated models for the design projects and noncorrespondence of the purposes of the studies to the actual requirements of the developers are significant. In particular, the basic "product" of the researchers is means of calculating quantitative parameters (numbers of devices, sizes of storage elements, and so on), that is, parametric optimization means, at the same time as the basic problems of the developers consist in selecting effective structures and algorithms for the operation (SAF) of the hardware and software, that is, structural optimization of the project. The SAF as the most conservative design elements must be selected in the early design phases and not be changed insofar as possible.

One of the most complex components of the KU with program control is the internal software (VPO). Analysis of a number of KU designs and publications [1] demonstrated that defining characteristics of the SAF of the VPO include the following:

И_1 - composition of the actuating processes (IP);

И_2 - configuration of the communications with respect to transmission of requests between the service objects and IP;

И_3 - configuration of the communications between the IP and the KU hardware;

И_4 - IP resource distribution algorithms;

Y_1 - composition of the control processes (UP);

Y_2 - configuration of communications for transfer of control between processes;

Y_3 - configuration of communications between the UP and the hardware;

Y_4 - UP algorithms.

By the process (IP and UP) we mean the part of the VPO which is initiated when transferring control with respect to some input.

The service objects ($И_2$) for the IP are subscribers, the KU equipment and operating personnel or signals (requests) detecting them, equipment elements (ferrodes, monitoring circuits, directive registers, and so on). For the UP, the "objects of service" are the IP and UP of lower level (Y_2).

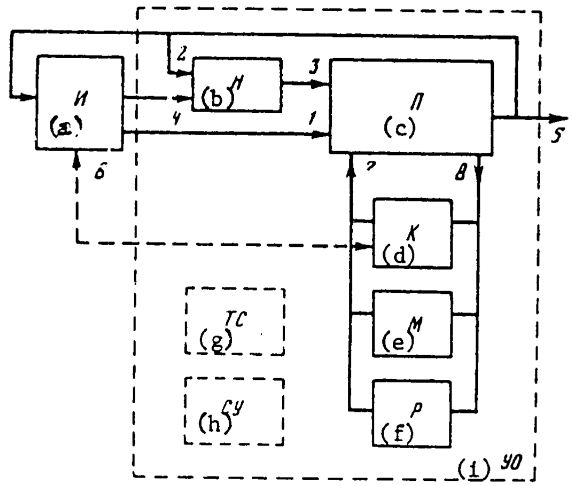
The hardware mentioned in the $И_3$ and Y_3 includes the control computer assemblies which provide for operation in the multiprogram mode (processors, interrupt systems, timers, and request storage elements).

Each of the enumerated VPO characteristics is defined in terms of the values of simpler quality characteristics (types of interrupt systems, timers, disciplines, and so on) and quantitative parameters (the dimensions of the storage elements, the priorities, and so on), which are elementary design solutions (EPR) for the developer. The directional sort of the EPR considering their combinability, researcher recommendations and, possibly, simulation results is the basic method of synthesizing the SAF VPO for the developer.

The schematic for the formation of the design solutions for constructing the VPO constructed on the basis of the morphological approach [2] confirms the extraordinarily high dimensionality of the problem of SAF synthesis and the necessity for decomposition of it -- isolation of partial problems of SAF analysis and synthesis and partial models of the object (ChMO) corresponding to them. However, the systems approach requires that in this case all the ChMO be representable in the form of reductions of some general model of the object (OMO), where reduction of the latter must be realized as a result of cutting out certain elements and detailing others and the OMO communications.

Since the control in the KU is realized as a set of request detection processes and dynamic distribution of the KU resources among them, only a queueing system (SNO) containing all types of sources of requests, resources and interactions between them which are characteristic of the KU as the object of simulation, can be used as the OMO. Analysis of the known classes of SMO [3] demonstrated that all of them are inadequate to the KU as the object of investigation, and the basic cause for this consists in the fact that between individual types of KU resources there are dominance ratios which do not exist in the known SMO classes. Therefore it is proposed that the SMO of the type presented in the figure, hereafter called SMO with nonuniform resource and dominance (SMO NRD), be used as the OMO.

The SMO consists of some number of sources (I) of requests and the service junction (UO), which, in turn, contains storage elements (N) and service devices of the following types: processors (P), sets (K), markers (M) and registers (R). The dominant devices are P which participate in servicing all the requests and for some of them realize search for a free and available device of the K, M or R type, using it (7) and release of it (8). The requests occurring in I are detected via the interrupt systems P or by scanning and they go to the UO directly (1) or through the N (4,3). The N are also used to store (2) the requests, during the servicing of which planned or random interrupts occur. Certain types of K are converted to I (6) when busy. Assignment of arbitrary numbers of the types of I, N, P, K, M, R and the quantities of elements of each type is possible.



Key:

- a. I
- b. N
- c. P
- d. K
- e. M
- f. R
- g. TS
- h. SU
- i. UO

The sequence of executing the IP with respect to single requests of each type, the procedure for use and release of the required UO resources, the rules for using the N and departure of the requests from the system (5) are given by the flow charts (TS) [4]. The control strategies (SU) give the resource distribution algorithms of the UO under load conditions and in accordance with the TS. The characteristics И_1-И_3 form the TS, and И_4, У_1-У_4, the SU.

In addition to the enumerated "large" components, the description of the OMO also contains groups of parameters characterizing the behavior of its individual elements. For example, for description of each group of uniform I, assignment of the following parameters is required: the number of I in the group, the number of requirements in one I, the sign of the fact that the I of the given group is formed from K elements when they are busy, the sign of activity (the request causes an interrupt or is detected during scanning of the I), the type of generation time distribution, type of impatience function, the characteristics of the indicated functions.

Thus, all types of service objects and KU resources (the switching bank is represented in terms of the availability parameters of the K), and also all of the above-enumerated characteristics of the SAF of the VPO have found their reflection in the given model. Consequently, the described SNO of the NRD is adequate to the KU as the object of investigation and the goals of the VPO developers.

The problem of the applicability of the proposed model is solved as follows.

1. Realization of the OMO in complete volume is meaningful only as a system of automating the design of high-level KU permitting the developer to compare elemental and complex design solutions matched with some system of forming them. The low recurrence rate of the solution of the synthesis problem of the SAF VPO makes this application of it highly problematic.

2. Realization of the ChMO containing all of the OMO elements is possible, but a limited number of versions of TS and SU, namely, the versions which describe the possibilities of the specific KU design system.

This realization, which is a simulation system with the dialog mode can be used for complex parametric adjustment of the installed KU for specific operating conditions.

3. Using the OMO jointly with the system for formation of the design solutions, by reduction of it it is possible to obtain the ChMO permitting realization in the form of analytical calculation schemes or simple simulation models describing the "elementary" processes in the KU. Considering through the OMO the influence of other processes on the investigated process, it is possible to obtain highly reliable recommendations with respect to efficient realization of it

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CONSIDERATION OF THE LENGTH OF TRANSMITTED MESSAGES FOR ROUTINGS IN CHANNEL-SWITCHED NETWORKS

[Article by N. I. Kuznetsov and O. N. Romanov, Moscow, pp 94-97]

Traditional routing algorithms in channel-switched networks (KK) do not consider the busy time of the channels when processing the next call inasmuch as this time, as a rule, is unknown when the call comes for processing. Another situation is observed when using the KK mode in computer communications networks where exchange is by messages of previously defined length and, consequently, the service time of the messages by the communications channels is known. It appears natural that consideration of the length of the transmitted messages (calls) and, consequently, the busy time of the channels processing them when solving the routing problems and the problems of load limitation can give an increase in operating efficiency of the network. For example, if the length of the sent message is small with respect to the average length of the messages in the network, it is obviously possible to send it over a bypass path; if the length of the message is long, this solution leads to inefficient use of resources and a reduction in output capacity of the network.

In reference [1] a very reasonable routing method is proposed which consists in the fact that for message transmission through the network the path with minimum cost is selected. The path cost is defined as the sum of the cost of the sides (the routings) on this path. The cost of a side is calculated as the mathematical expectation of the number of failures in this side as a result of the fact that the given message is received for servicing:

$$d_i' = \int_0^\infty \lambda\left[p_n(n,t/i+1) - p_n(n,t/i)\right]dt, \qquad (1)$$

where λ^0 is the intensity of the traffic flow to the side;

n is the number of channels in the side;

$p_n(k,g/i)$ is the probability that k channels on the routing will be busy at the time t under the condition that at the initial point in time i channels were busy.

Let us consider the routing as a multiple-queue queueing system. We shall consider that the lengths of the messages reaching the routing and, consequently, their servicing times in the channels are distributed by an exponential law. Under the assumption of the simplest traffic flow to the side, the probabilities

$p_n(k,t,i)$ are solutions of the corresponding system of Kolmogorov-Chapman ordinary differential equations.

Below, the authors have proposed a generalization of formula (1) to two cases of considering the length of the transmitted messages when solving the routing problem.

Let us determine the cost of transmitting a message of length ℓ over a routing of n channels under the condition that at the time t, i channels are busy for a random time distributed by an exponential law as

$$d_i''(\ell) = \int_0^\infty \lambda[\tilde{p}_n(n,t/i+1,\ell) - p_n(n,t/i)]\,dt, \qquad (2)$$

where $\tilde{p}_n(n,t/i+1,\ell)$ is the probability that all n channels will be busy at the time t under the condition that at the initial time i channels were busy, and the (i+1) channel is busy for the time ℓ.

It is easy to reduce formula (2) to the form

$$d_i''(\ell) = \lambda\int_0^\ell [p_{n-1}(n-1,t/i) - p_n(n,t/i)]\,dt + \lambda\int_\ell^\infty \Big[\sum_{j=0}^{n-1} p_{n-1}(j,\ell/i)\cdot p_n(n,t-\ell/j) - p_n(n,t/i)\Big]dt, \qquad (2a)$$

where all the probabilities are solutions of the corresponding Kolmogorov-Chapman systems. It is possible to obtain the value of $d_i''(\ell)$ in explicit form for a specific value by solving the corresponding Kolmogorov-Chapman systems and further integration. Figure 1, a, b, c shows the form of these functions for the cases n=1,2,3.

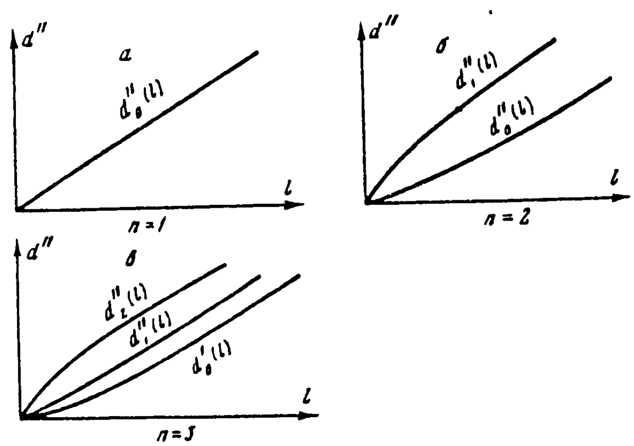

Figure 1

For $\ell\to\infty$ the function $d''_i(\ell)$ approaches an asymptote, the slope and the displacement of which depend on the flow intensity. The slope is identical for all i.

In the case where at the time of appearance of a message of length ℓ the times $b_1, b_2, \dots, b_i$ of completion of servicing of the calls using busy channels are known, formula (1) assumes the form:

$$d_i'''(\ell, b_1, b_2, \dots, b_i) = \int_0^\infty \lambda\left[p_n(t/i+1, b_1, \dots, b_i, \ell) - p_n(t/i, b_1, \dots, b_i)\right]dt, \tag{3}$$

where the variable of the type $p_n(t/j, b_1, \dots, b_j)$ is the probability that all n channels in the side will be busy under the condition that at the initial point in time j channels out of n were busy for the times $b_1, \dots, b_j$.

The values of d_i''', just as d_i'', can be obtained in explicit form for the specific values of the parameters entering into formula (3). Figure 2 shows the form of the function d''' for n=2.

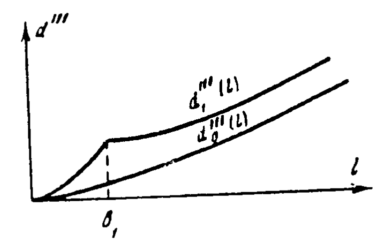

Figure 2

It is possible to introduce the average cost of messages serviced by the network into formulas (1), (2), (3). In this case it is possible to consider d'_i, d''_i, d'''_i as the total cost of the messages which will be rejected as a result of servicing the given call. This permits use of not only the number of lost calls, but also their total length as the purpose function, which has great significance when optimizing the volume of information transmitted by the network. In the general case the message costs and the purpose function can be arbitrary.

It is necessary especially to note that the discussed approach permits solution of both the routing and the load limitation problems. Actually, when determining the message transmission routing, a path is selected with minimum cost, that is, with minimum losses. If the expected losses are greater than the cost of the transmitted message itself, then it is rejected and, thus, the load limitation problem is solved. The authors performed a statistical simulation of a network with KK for the three enumerated methods of determining the cost of sending a message over the channels. For simplification of the statistical model it was proposed that the intensities of the level of messages in all channels are equal and commensurate with the intensity of the calls in the network as a whole.

The operating efficiency of the network was estimated by the reject coefficient (K_{reject}) which is the ratio of the number of messages which will be rejected to the total number of messages reaching the network, and the loss coefficient (K_{loss}) which is the ratio of the sum of the lengths of the messages which were rejected to the total length of all messages.

The simulation demonstrated that in the case where the cost of sending a message was determined in accordance with formula (2), K_{reject} and K_{loss} decreased by 15-20% and 5-10%, respectively, by comparison with the case where the cost of sending was calculated in accordance with formula (1), that is, when the message length was not taken into account at all when determining the cost of sending it through the communications channels.

Consideration of the time required to complete servicing of the messages which are already being serviced on the routing at the time of arrival of a new call (calculation of cost in accordance with formula (3)), further reduces K_{reject} and K_{loss} by 15-20% and 5-10%, respectively.

The representation of the mathematical expectation of a routing as a multiqueue queueing system determines the formulas for calculating the cost of sending messages in the network and the load limitation criteria. The performed statistical simulation confirms the correctness of the approach consisting in considering the lengths of the sent messages to the solution of the routing and load limitation problems.

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INFLUENCE OF CONTROL FUNCTION DISTRIBUTION ON OUTPUT CAPACITY OF A MULTIPROCESSOR CONTROL COMPUTER

[Article by S. Sh. Kutbitdinov, Leningrad, pp 97-100]

One of trends noted in the construction of the control systems of switching centers with programmed control is use of multiprocessor control computers (EUM) consisting of two or more processors having a common memory.

In reference [1] a study is made of the basic versions of the structural organization of two-processor EUM. A synthesis is made of the structural organization of the two-processor EUM by the criterion of maximum output capacity, by which we mean the maximum number of calls which can be serviced by the control system per unit time with given service quality.

The quality of servicing a call is determined by the probability of its loss and will be within the admissible limits if the probability P_{loss} does not exceed a given value, that is, if the condition $P_{loss} \leq P_{ad}$. Therefore in accordance with the definition of output capacity, it was demonstrated that

$$\Pi(P_{\text{пот}}) = (1 - P_{\text{пот}})\lambda, \quad (1)$$

Key: 1. loss

where λ is the intensity of the arrival of calls at the switching center.

For each of the investigated versions of structural organization of a two-processor EUM, expressions were obtained for the probability P_{loss} under the assumption that the scanning functions of the line, patch cord and order circuit systems are not realized by processors making up the two-processor EUM.

As a result of the performed analysis the best versions with respect to output capacity turned out to be the versions of structural organization of a two-processor EUM operating in the time-sharing mode for the load and in the function-sharing mode.

The choice of the two-processor EUM can be based on the fact that this type of control system is quite widely used in practice. The two-processor configurations are elementary structural parts from which significantly more complex multiprocessor structures can be assembled.

In the given paper a study is made of the influence of the distribution of the control functions on the output capacity of the two-processor EUM operating in the function-sharing mode, for which the functions realized in this phase of servicing the call are divided into two groups. Each group of functions is realized by one of the EUM processors; therefore the servicing of a call consists of two phases. The influence of the call processing function distribution between the two EUM processors on the output capacity of the entire control system is investigated considering performance of the functions of scanning the line, patch cord and order circuit systems by the processor of the first call service phase.

The mathematical model of the investigated version of the structural organization of a two-processor EUM can be represented in the form of a single-queue two-phase queueing system (SMO) with simplest input flow of requests of intensity $\Lambda=N\lambda$ (N is the number of call servicing steps), with unlimited queue ahead of the second phase and exponential service time distribution in each phase with intensity μ.

When servicing the call, part of the speed (ν) of the EUM will be spent on performing the scanning operations; therefore the intensity of servicing the calls during processing can be represented in the form

$$\mu_{o\delta}^{(1)} = \frac{\nu - K_{cc}^{(2)}}{K_{o\delta}},$$

Key: 1. processing; 2. scanning

where K_{scan} is the average number of operations per unit time spent by the EUM on scanning the line, patch cord and order circuit systems; $K_{processing}$ is the average number of operations spent by the EUM on processing one call.

For the investigated two-phase SMO, $K_{scan}=K'_{scan}+K''_{scan}$, but since it was noted that the scanning functions are performed only by the first phase processor, $K''_{scan}=0$.

For processing a call $K_{process}=K'_{process}+K''_{process}$, where $K'_{process}$ is the average number of operations spent on processing one call by the first phase processor; $K''_{process}$ is the number of operations spent on processing the same call by the second phase processor. The values of $K''_{process}$ and $K''_{process}$ are related to the value of $K_{process}$ by the expressions:

$$K_{o\delta}^{(1)} = \alpha K_{o\delta}, \quad K''_{o\delta} = (1-\alpha) K_{o\delta},$$

Key: 1. process

where α is a coefficient giving the distribution of the call processing functions between the two servicing phases (processors) $(0<\alpha<1)$.

For the described two-phase SMO, the expression was obtained for the call loss probability P_{loss} in the form:

$$P^{(2)}_{\text{пот}} = e^{-(A-\Lambda)T^{(1)}_{\text{доп}}} - \frac{A-\Lambda}{A-B} e^{-(B-\Lambda)T_{\text{доп}}} \left[e^{-(\Lambda-B)T_{\text{доп}}} - 1\right],$$

Key: 1. ad; 2. loss

where

$$A = \frac{\Gamma}{1-\alpha}, \quad B = \frac{\Gamma}{\alpha} - \frac{K_{\text{ск}}^{(1)}}{\alpha K_{\text{од}}^{(2)}}$$

Key: 1. scan; 2. process

T_{ad} is the admissible time the request spends in an individual call servicing step.

From the obtained expression it is obvious that for a fixed value of K_{scan}, by varying the value of α it is possible to obtain different values of P_{loss}. Using the initial data close to the actual data, a numerical calculation was performed, and the functions $P_{loss}(\alpha)$ were constructed for fixed values of K_{scan}.

Investigating the influence of the value of α on P_{loss} for a fixed value of K_{scan}, an optimal value of (α^*) was found for which $P_{loss}(\alpha^*)=\min P_{loss}(\alpha)$, that is, the value of $\alpha^*(K_{scan})$ determining the optimal distribution of the call processing functions between the EUM processors.

Thus, from what has been stated it is possible to draw the following conclusion.

1. Great achievements in the field of microprocessor engineering have led to the appearance of distributed and mixed methods of program control in the switching centers for which the performance of individual call processing functions is invested in individual microprocessors or specialized processors. Therefore it is necessary to develop a procedure for optimal distribution of the call processing functions among several processors to increase the output capacity of such control systems.

2. The performed study of the operation of a two-processor EUM operating in the function-sharing mode demonstrated that the distribution of the call processing functions among processors greatly influences the output capacity of the entire EUM.

3. The developed method of optimal distribution of the call processing functions among processors can be extended to multiprocessor EUM which contain more than two processors with identical speed having a common memory.

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ROUTING ALGORITHMS AND COMMUNICATIONS QUALITY IN A MULTIPOLAR DATA TRANSMISSION NETWORK

[Article by N. P. Krutyakova, Leningrad, pp 100-103]

For breakdown of the overall problem of designing an optimal multipolar communications network it is possible to isolate the problems of synthesizing the network control in an independent class [1]. The algorithms for the solution of these problems can make up the software for the network control centers during operation of the network and also be used in the automated communications network design system and the network control system.

The problems of synthesizing the communications network control in general and data transmission network (PD) in particular, belong to the class of mathematical programming problems. Very frequently the probability of timely delivery or quality of communications Q is used as the purpose function in them. The form of the purpose function Q of the network essentially depends on the traffic flow routing algorithm. Thus, for strict solution of the problem of control synthesis it is first necessary to solve the analysis problem: to determine the analytical expression Q as a function of the load characteristics of the network, its structural parameters and routing algorithm. For this purpose, a study was made of a tripolar network as the elementary cell of a multipolar PD [data transmission] network, the more so in that it is possible to propose structural stability of the results of investigating such a network and extension of these results to networks with a large number of junctions [2].

Three routing algorithms were considered: message distribution by fixed routes (1), a set of fixed routes with probabilities $\{p_{ijq}\}$ where p_{ijq} is the probability of transmission of messages over the q-th path from the existing $\mathcal{L}_{ij}$ paths between i and j junctions of the network (2), the distribution of messages using bypass routings when one basic and several bypass routings are defined for each message flow λ_{ij} in the network, but the bypass routings are used if it is impossible to use the basic routing (3). The basic assumptions are as follows: the message flows in the network are Poisson; the switching centers (UK) and communication branches (VS) are absolutely reliable; the information aging time and the message service time are distributed by an exponential law with intensities ν_i, μ_i, respectively, for the i-th VS; the service type in the VS is with rejects (O), limited waiting (OOZh), and unlimited waiting (OZh); the messages serviced through a tandem junction in the network with rejects occupy two communications channels simultaneously (for a tripolar network); here

$C_{e\ell k\ell}=C_{e\ell ijs}$, s=1,2, [3], and, consequently, $\mu_{k\ell}=\min_s \mu_{ijs}$, s=1,2, where $C_{e\ell ij}$ is the operating carrying capacity of the VS between the i-th and j-th junctions of the network, $i,j\in\{1,\ldots N\}$, N is the number of network junctions. As has already been noted, the problem consists in determining the analytical expressions for the communications quality Q for a network with the same topology, but different network algorithms for servicing the messages or routing algorithms.

The assumptions made with respect to the nature of the processes occurring in the network permit consideration of it as a Markov type queueing system. The method of solving the problem consists in constructing phase spaces of the Markov model of the network for different routing algorithms and finding the probability vector of different states of the network under steady-state conditions, $P=(P_i)$, $i=\overline{1,k}$ determined from the equation P=PW [4], where P_i is the steady-state probability of the i-th state of the Markov model of the network; k is the number of states of the Markov model of the network; W is the matrix of transition probabilities of states of the network. The transition matrix elements W are calculated by known procedures for a queueing system [4, 5]. Q is defined after finding the vector P. Thus, for the PD network with rejects when N=3, $\nu_i=0$, $i=\overline{1,3}$, the communications quality Q for the following methods of message servicing was determined:

a) The message distribution over the direct (shortest) routings between network junctions (algorithm 1):

$$Q_1'=\sum_{i=1}^{3}\frac{\lambda_i}{\lambda_\Sigma}Q_i\text{, where }\lambda_\Sigma=\sum_{i=1}^{3}\lambda_i,\; Q_i=\frac{\mu_i}{\mu_i+\lambda_i},\; i=\overline{1,3},$$

λ_i is the message flow intensity in the i-th VS.

For a uniform network when $\lambda_i=\lambda$, $\mu_i=\mu$, $i=\overline{1,3}$,

$$Q_{10}'=\frac{\mu}{\mu+\lambda}$$

b) The message distribution when using a tandem routing for one of the flows λ_i, $i\in\{1,2,3\}$ (algorithm 1):

$$Q_i''=\frac{1}{1+\frac{\lambda_i}{\mu_j}\prod\limits_{\substack{k=k',\\k'\neq i}}Q_k}\left[\sum_{\substack{k=k',\\k'\neq i}}\frac{\lambda_k}{\lambda_\Sigma}Q_k+\frac{\lambda_i}{\lambda_\Sigma}\prod_{\substack{k=k',\\k'\neq i}}Q_k\right],$$

where $\mu_j=\min\{\mu_j'\}$, μ_j', $j=1,2$, is the message service intensity in the corresponding branch entering into the tandem routing.

For a uniform network

$$Q_{10}''=\frac{3\mu\lambda+3\mu^2}{3\lambda^2+9\mu\lambda+3\mu^2},$$

c) The message distribution when using probability distribution of one of the flows λ_i, $i \in \{1,2,3\}$ over possible routings (algorithm 2):

$$Q_2 = \frac{1}{\lambda}\Big[\sum_{\substack{k=k',\\ k'\neq i}} \frac{\lambda_k}{\lambda_\Sigma} Q_k + \frac{x\lambda_i}{\lambda_\Sigma} Q_i(x\lambda_i)\cdot A + \frac{(1-x)\lambda_i}{\lambda_\Sigma} \prod_{\substack{k=k'\\ k'\neq i}} Q_k\Big],$$

where

$$A = 1 + \frac{(1-x)\lambda_i}{\mu_j} \prod_{\substack{k=k',\\ k'\neq i}} Q_k ;$$

x is the part of the messages of the flow λ_i serviced over the direct routing.

For a uniform network

$$Q_{20} = \frac{1}{3}\left(\frac{2\mu(\mu+\lambda)+(1-x)\mu^2}{(\mu+\lambda)^2+(1-x)\mu\lambda} + \frac{x\mu}{\mu+\lambda}\right);$$

d) The message distribution when using bypasses for one of the flows λ_i, $i \in \{1,2,3\}$ (algorithm 3):

$$Q_3 = \frac{1}{\lambda'}\Big[\sum_{\substack{k=k',\\ k'\neq i}} \frac{\lambda_k}{\lambda_\Sigma} Q_k + \frac{\lambda_i}{\lambda_\Sigma} Q_i \cdot A' + \frac{A'-1}{\lambda_\Sigma} \sum_{\substack{k=k',\\ k'\neq i}} \lambda_k\Big],$$

where

$$A' = 1 + \frac{\mu_j + \lambda_i}{\mu_j + \mu_i + \lambda_i} \cdot \frac{\lambda_i}{\mu_j} \prod_{\substack{k=k'\\ k\neq i}} Q_k .$$

For a uniform network when $\mu_i = \mu = 1, \lambda_i = \lambda$, $i = \overline{1,3}$

$$Q_{30} = \frac{\frac{5}{3}\lambda^2 + 4\lambda + 2}{\lambda^3 + 5\lambda^2 + 6\lambda + 2}.$$

For the network with $N > \phi$ and $\nu_{ij} \neq 0$, $i,j \in \{1,\ldots N\}$, Q was determined when using a Markov model of the network with absorbing states. In this case Q of the network is represented as a function Q of the VS and the routing algorithm.

Studies of the network Q as a function of λ_i/μ_i made it possible to discover the advantages of one routing algorithm or another for different network loads and also the behavior of the functional Q. Thus, for example, the studies demonstrated that $Q_3 \geq Q'_1$ in the entire investigated range of low variation. This is also confirmed analytically. In [2] it was determined that for $\lambda > 1$ in a uniform network the communications quality when using bypass paths for all flows is less than Q'_{10}, that is, there are limiting loads in a tripolar uniform network for which algorithm has has no advantage. Considering what has been discussed, it is obviously possible to talk about the maximum number of flows in a nonuniform network, designation of bypass routings for which retains the advantages of algorithm 3 in the entire range of load variation.

The solutions of the enumerated analysis problems permit transition to the solution of the problems of synthesizing the control algorithm by the criterion of max Q. In particular, exact analytical expressions are obtained for the parameters of algorithm 2 in a tripolar network where a defined ratio of network parameters was obtained $\sqrt{Q_1 Q_2 (Q_1 - (\lambda_2/\mu_1) Q_2)} \geq Q_3$, only on the satisfaction of which is it expedient to use algorithm 2 as compared to algorithm 1.

The obtained expressions for Q and their analysis permit exact solution of the problem of message flow control synthesis or determination of the class of optimal control algorithms for the network structure and message flows for various load characteristics of the network in order to insure max Q.

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STRUCTURAL PRINCIPLES OF AN AUTOMATED DESIGN SYSTEM FOR INFORMATION DISTRIBUTION SYSTEMS AND DEVICES

[Article by V. G. Lazarev, N. Ya. Parshenkov and Ye. I. Piyl', Moscow , pp 104-106]

[Text] Progress in the field of communications, computer engineering and microelectronics has led to significant development of information distribution systems. The information distribution systems (communication systems) represent an example of complex systems, and they include the channels and channel-forming equipment, information transmission interception means, switching equipment and other devices providing for the transmission and distribution of various information among the sources and receivers (users).

At the present time computer engineering means are finding broad application in communication systems. This is leading to a qualitative change in the structural principles of both the communication networks and the control systems in the communication centers and networks. A new system with programmed control has come to replace the crossbar switching centers (UK) with register-marker control. In the new system the processes of servicing the requests to set up calls are controlled by programmed devices, the nucleus of which at the medium- and large-capacity UK is made up of control computers (EUM) analogous with respect to structural principle, operation and capabilities to the most modern computers. At the low-capacity UK, microcomputers and special control units (UU) in the form of microprogrammed UU are finding broad application.

The UK with programmed control create good prerequisites for accelerating the introduction of dynamic control of information flows on the communications networks ensuring a significant increase in the efficiency of the use of the channel and switching equipment by comparison with the presently used static information distribution as a result of redistribution of the flows with a change in situation on the communication network. Thus, at the present time the control system is an independent, quite highly organized element of the communication system based on broad application of complex control devices and modules. Therefore the design of the control system as a whole and its individual devices is becoming such a labor-intensive, complex process that it does not appear possible to construct a sufficiently efficient control system for a modern communication network manually, using only design experience and intuition.

The design of other elements of a modern communication system (the structure of the communication network with selection of the channel group capacities and/or their carrying capacities, the structure of the switching system of the UK) also requires the application of computers.

Consequently, at the present time it has become necessary to build an automated design system for information distribution systems and devices (ASPRI).

When building the ASPRI, undoubtedly it is necessary to consider the existing experience of a number of organizations with respect to developing automated design systems (ASP) for individual elements of the communication system (the automated design system for network structure, the automated system for calculating call losses in the switching system, and so on).

However, in our opinion it is necessary to create not individual automated systems for designing and calculating the parameters of the network elements or UK, but an integrated ASP constructed on a united procedural base. The creation of the integrated ASPRI even with staged design of the communication system will permit consideration of the interrelation of its individual elements. The structural principles of the integrated ASPRI during the development of which research experience at the IPPI of the USSR Academy of Sciences with respect to automation of the design of individual elements of a communication system are discussed.

Considering the properties of the communication system and the capabilities of modern computers, it is expedient to construct the ASPRI in the form of a multimodular hierarchical structure. Here the ASPRI is an open system and permits inclusion and/or replacement of its individual functional modules (subsystems). The structural principles of the ASPRI and its individual subsystems have been developed as applied to the channel switching network. However, the ASPRI permits inclusion of the required subsystems and modules for consideration of the specific nature of the design of the message and packet switching networks.

The ASPRI includes four subsystems: the design of the communication network, the selection of the optimal flow control algorithms, the selection of the architecture of the control system and design of the EUM.

The ASPRI operation system is a four-level hierarchical structure. The first (highest) level includes the chief monitor of the system, the control and service programs providing for the choice of one of the subsystems in the dialogue mode. On the second level (the subsystem level) there are subsystem monitors and control programs for the subsystem providing for the selection of functional modules in the dialogue mode permitting the design of quite powerful devices of the communication system (for example, the design of the communication network topology, the microprogram UU, and so on). The monitor and the control programs of the functional module providing for selection of the functional modules are located on the third level (the functional module level).

Each of the functional modules designed for solving a specific problem (encoding the internal states of the automaton, selection of the channel distribution method, and so on), has a monitor and fourth-level control programs which provide for inclusion and operation of defined functional design programs.

The given ASPRI does not impose special requirements on the requester with respect to knowing formalized languages or design methods. The requester communicates with the ASPRI in Russian, using terms and concepts familiar to him. Under these conditions, for simplification of the operation system and the translators, the "active designer-passive requester" principle is used in the ASPRI.

The developed structural principles of ASPRI permit broad use of external storage elements and archives.

The resident part of the operation system of the ASPRI is only the chief monitor which takes a small amount of the ready-access memory of the computer.

INTRODUCTION OF A METHOD OF SETTING UP CALLS WITH ALTERNATIVE ROUTINGS ON RURAL TELEPHONE NETWORKS

[Article by Yu. V. Lazarev and S. A. Krasnov, Moscow, Rostov , pp 106-108]

[Text] As is demonstrated in a number of papers [1, 2], one of the possible ways to improve communications quality and reliability is the introduction of a method of setting up calls with bypass routings combined with dynamic distribution of the call flows. However, up to now the structure of the rural telephone networks (STS) provided for by the process design norms did not permit the possibility of introducing the bypass method of establishing connections, for the STS was constructed by the radial-nodal principle. In addition, ATS [automatic telephone offices], the control units of which did not have the possibility of implementing the method of setting up calls with bypasses, were used in the STS.

At the present time, in the presence of significant gravitation between ATC in the STS the Process Design Norms provide for the organization of transverse couplings between the ATS of the same level, which, in turn, permits organization of bypass routings from the point of view of the construction of the STS. However, the equipment of the ATS widely used at the present time on the STS (ATSK 100/2000, ATSK 50/200), with the exception of the ATSK 50/200 M, does not permit organization of the method of setting up calls with bypass routings on the STS.

During studies of the efficiency of introducing the method of setting up calls with bypasses and dynamic control, more attention was given to the long-distance (MTS) and municipal telephone networks (GTS) than the rural telephone networks which can be explained by the restrictions imposed by the structure of the STS and the ATS used in the STS on the possibility of introducing a bypass method of setting up calls.

In the example of one of the STS, a study is made of the problem of the expediency of introducing the method of setting up calls with bypasses and dynamic call traffic distribution on the rural telephone networks.

Results are presented from observations of the load variation between individual automatic telephone offices. The main conclusions which can be reached on the basis of the measurement data consist in the following:

a. during the day the load is subject to significant fluctuations;

b. the peak load hours (plh) on individual service routings do not coincide with respect to time of day;

c. the time of occurrence of the plh on individual service routings remains invariant for different days of the week;

d. the increase in load on the STS begins at 0800 hours, and the decrease in the traffic is observed beginning at 1700 hours;

e. the nature of the load variations on individual service routings basically depends on the nature of the production activity of the subscribers included in the automatic telephone offices between which the given routing is organized.

In order to discover the possibility of introducing the method of setting up calls with bypasses and dynamic call traffic distribution in the investigated STS, on the basis of the obtained loads, a check calculation was made of the quality of servicing the calls. The calculation was performed on the basis of a method, the description of which is presented in [3]. Here the calculation was performed for three cases: for the method of setting up calls without bypasses, for the method of setting up calls with bypasses and for dynamic distribution of the call traffic based on the method of limiting the tandem load during overloads when a threshold is established for the tandem traffic in the form of the number of busy channels, on exceeding of which it is forbidden to send tandem calls over the given routing, that is, it is forbidden to use the given routing as a bypass routing.

The given check calculation demonstrated that with the introduction of the method of setting up calls with bypasses, noticeable improvement in the quality of servicing the calls is observed. In addition, analysis of the obtained results demonstrated that a still more noticeable effect in improving the quality of servicing the calls will be observed on introducing dynamic distribution of the call traffic.

During the process of performing the check calculation, a section of the STS was discovered in which the losses with the method of setting up calls without bypasses at certain points in time significantly exceeded the normative values of the losses.

With the introduction of the method of setting up calls with bypasses, the losses between the automatic telephone offices in the given section, although they decreased, still remained above the losses provided for by the norms. On introduction of dynamic call traffic distribution on the basis of the methods of restricting the tandem load during overloads, the losses obtained during the calculation process turned out to be lower than the normative losses.

In order to organize the method of setting up calls with bypasses, a correction was made to the automatic telephone office equipment used in the investigated STS.

In particular, a module for organizing bypasses was introduced into the outgoing stage markers, by means of which the calls not traveling over the direct path (the first choice path) because of all the channels being busy in this group were not lost, but were directed to the bypass routing (the second choice path).

Considering certain difficulties in introducing a completely automatic system of dynamic call traffic distribution on the STS in view of the presence of a crossbar-type automatic telephone office on the network, the dynamic distribution of the call traffic was organized by a schedule considering the plh on individual service routings.

By calculating the quality of servicing the calls, the optimal schedule was selected for limiting the tandem flow of calls for which the losses obtained at any point in time are below the normative values of the losses.

When selecting the schedule, the method of selecting the optimal call flow distribution plan was used [4].

Then an experiment was run on the investigated section of the STS which demonstrated that on introduction of the method of setting up calls with bypasses in combination with the dynamic call flow distribution elements, the call losses between the automatic telephone offices decreased.

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INFLUENCE OF THE CARRYING CAPACITY OF SWITCHING CENTERS ON DYNAMIC CONTROL EFFICIENCY

[Article by Yu. V. Lazarev and I. V. Nikiforova, Moscow , pp 109-111]

[Text] One of the means of improving the use of communication channels and lines and at the same time more efficient use of the communication network structures is the introduction of dynamic call traffic distribution. However, as was demonstrated in a number of papers, for example, in [1, 2], depending on the structural principles of the communication network, its operating conditions, various methods of dynamic control can give different effects. Thus, in [2] it was demonstrated that the game method of dynamic control as applied to the structural principle and operating conditions of the city telephone networks (GTS) is preferable over, for example, the relief method. This is explained by the fact that considering the peculiarity of the structure of the GTS (significant connectedness of the switching centers and restrictions with respect to losses), it is expedient to use only the paths which pass through one tandem switching center (UK) as the bypass paths. Under these conditions, for the relief method the criterion for selecting the path of setting up calls for which the path length expressed in the number of tandem sections is used, all of the bypass routings will be equivalent. By the game method, the criterion for selecting the path of setting up calls, is the probability of setting up the call. Accordingly, in the game method the bypass routings for setting up calls will not be equivalent, and they are classified as a function of the probability of setting up the call.

It must be noted that up to now the efficiency of introducing one method or another of dynamic control has been investigated without considering the carrying capacity of the UK, at the same time as it is possible to expect that the carrying capacity of the UK will have significant influence on the efficiency of introducing various dynamic control methods. In order to determine the influence of the UK carrying capacity on the effectiveness of introducing one dynamic control method or another, a number of GTS networks have been simulated in which the structural principles and operating features of the GTS were taken into account. For comparison a study was made of the static distribution of the call flows and two dynamic control methods: the relief method and the game method.

The results of simulating the GTS networks are presented which confirm the assumption made previously that the carrying capacity of the UK has significant influence on the effectiveness of introducing various dynamic control methods.

Analyzing the results obtained, it is possible to draw the following conclusions:

a. introduction of the relief method improves the quality of call servicing insignificantly by comparison with the static call traffic distribution. Introduction of the game method leads to more noticeable improvement of the call servicing quality by comparison with static distribution than the relief method;

b. with a decrease in carrying capacity of the UK, the effect from introducing the dynamic control methods decreases;

c. on introduction of different carrying capacity for different UK, a significant difference is observed in the results obtained for the game method and the relief method. Here the average network losses obtained for the game method are less than the average network losses obtained for the relief method.

The fact of a decrease in the effect from introducing dynamic control methods with a decreasing carrying capacity is obvious in general. Decreasing the carrying capacity of the UK for tandem call traffic, at the same time we exclude the possibility of using bypass routings which, in the final analysis, also has a negative effect on the call servicing quality.

The fact that for a different carrying capacity of different UK with the game method of dynamic control significantly smaller average network losses are obtained than for the relief method is explained as follows. As has already been pointed out above, the path selection criterion for establishing the connection in the relief method is the path length expressed in the magnitude of the tandem sections. From the point of view of the given criterion all of the bypass routings for setting up calls are equivalent, although these routings differ significantly with respect to carrying capacity. The difference in carrying capacity is explained by the fact that each bypass routing consists of three sections (channel group, tandem UK, channel group) characterized by their carrying capacity, and the carrying capacity of the entire route will be characterized by the least of the carrying capacities making up the given bypass route for setting up the call. Thus, if the carrying capacity of the channel routes is greater than the carrying capacity of the UK, with the relief method a nonoptimal plan for distribution of the call traffic can be formed.

By the game method, the criterion for which is the probability of setting up the call, differences will be discovered in the carrying capacity of different bypass routings and calls will be directed last to the bypass routings having the least carrying capacity. The obtained results permit the conclusion to be drawn that when estimating the efficiency of introducing one method of dynamic control or another, it is also necessary to consider the carrying capacity of the UK. Failure to consider this factor can lead to a significant error when estimating the effectiveness of introducing one method of dynamic control or another.

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APPLICATION OF THE METHODS OF DYNAMICS OF MEANS FOR ESTIMATING THE EFFICIENCY OF DYNAMIC TRAFFIC CONTROL IN NETWORKS WITH QUEUES

[Article by Yu. A. Lev, Kiev , pp 111-113]

[Text] In [1], a study was made of the possibility of using the method of dynamics of means [2] for analyzing networks with queues. The discussion is conducted using the example of a message-switched (KS) network. The problem reduces to solving a system of differential equations with respect to the number of messages in different parts of the network, that is, the lengths of the message queues. Here the primary problem is determination of the system coefficients in each step of solving the system of differential equations for the investigated control algorithm. The coefficients are the message traffic intensities directed to each branch of the network, the flows of branch releases, and the outgoing message flows from the network. A procedure is proposed for determining the indicated values.

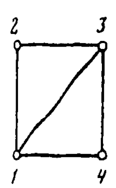

Figure 1.

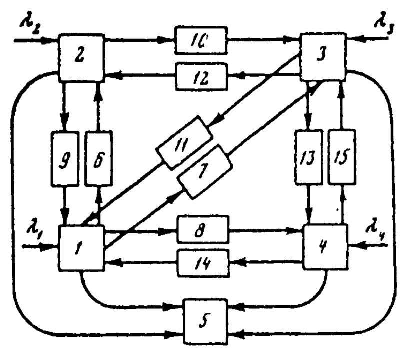

Figure 2.

Let us consider the example of a KS network depicted in Figure 1. Figure 2 shows the graph of the states of the message circulating in the network. The numbers 1 to 4 denote the incoming states of the message to the corresponding network junction, the number 5 denotes the outgoing state of the message from the network. The numbers 6-15 denote the states of the message in the queue and being serviced in the corresponding branches of the network.

Let us make the following assumptions. The external traffic flows of the network are Poisson. The message lengths are distributed by an exponential law and are determined independently in each junction (the hypothesis of Kleinrock independence). The flows of releases of the branches are Poisson with intensities equal to the corresponding intensities of the input flows of the branches if the branch load is less than one (the Bern theorem). Otherwise the flow of branch releases is Poisson with intensity equal to the carrying capacity of the branch. All of the partial flows running through the given branch are limited to an equal degree. Thus, all the processes occurring in the system described by the graph (see Figure 2) are Markov. The system of equations for the average numbers of states of the message (average lengths of queues) look like the following:

$$\begin{aligned}
\frac{dm_1}{dt} &= \lambda_1 + \lambda_{9.1} m_9 + \lambda_{11.1} m_{11} + \lambda_{14.1} m_{14} - \\
&- \lambda_{1.6} m_1 - \lambda_{1.7} m_1 - \lambda_{1.8} m_1 - \lambda_{1.5} m_1; \\
&\dots\dots\dots \\
\frac{dm_5}{dt} &= \lambda_{1.5} m_1 + \lambda_{2.5} m_2 + \lambda_{3.5} m_3 + \lambda_{4.5} m_4; \\
\frac{dm_6}{dt} &= \lambda_{1.6} m_1 - \lambda_{6.2} m_6; \\
&\dots\dots\dots \\
\frac{dm_{15}}{dt} &= \lambda_{4.15} m_4 - \lambda_{15.3} m_{15},
\end{aligned} \tag{1}$$

where λ is the intensities of the external loads of the flows of message routings for servicing to the corresponding branch, the flows of releases of the branches and exit of the messages from the network; m are the mean numbers of states of the message.

All the coefficients of the system (1) are defined for a fixed plan of traffic distribution when calculating the right-hand sides of system (1). The location of the input flow intensities λ_{inp} and output flow intensities λ_{out} for each branch which are related by an expression of the following type is basic:

$$\lambda_{out} = \alpha\lambda_{inp}; \quad \alpha \le 1.$$

The remaining coefficients are defined in terms of λ_{inp} and λ_{out} for each branch in terms of the traffic distribution plan. The values of λ_{inp} for each branch are defined as the sum of the intensities of partial flows coming into each branch considering the flow distribution plan. The values of α are found by solving the following system of linear algebraic equations:

$$\alpha_i = C_i/\lambda_{inp_i}; \quad i = 1, \dots, M, \tag{2}$$

where C_i is the carrying capacity of the branch i; M is the number of network branches; if $C_i/\lambda_{inp_i} > 1$, then $\alpha_i = 1$.

The solution of system (1) in one step takes place for a fixed traffic distribution plan. On transition to the next step of the solution of the system (1) a correction is made to the flow distribution plan if necessary using the investigated flow control algorithm. Here it is mandatory to use an algorithm to eliminate possible loops in the routings or the algorithm to get out of the loops, for when solving system (2), each flow is taken from the source center to the addressee center; on occurrence of a loop in the routing the program for solving system (2) will loop. Any method can be used to solve systems (1), (2). Here, in addition to the values of the mean lengths of queues in the branches, the mean values of the total lengths of queues throughout the entire routing can be obtained and, consequently, the values of the mean message delays for each connection.

The application of the method of dynamics of means for analysis of the networks was checked in an example (see Figure 1). The program was written in Fortran for the BESM-6 computer. For system (2), the iteration method turned out to be convenient from the algorithmic point of view. For solving system (1), a standard BESM-6 program was used. The preliminary results indicate high speed of the method.

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CALL DISTRIBUTION ALGORITHM ON PROSPECTIVE RURAL TELEPHONE NETWORKS

[Article by I. O. Litsit, Riga , pp 113-117]

[Text] In recent years equipment systems have appeared for quasi-electronic rural automatic telephone offices (ATS KE) for operation with the existing automatic telephone offices in existing communications networks. Digital transmission systems with pulse-code modulation (PCM) and also medium-speed data transmission equipment for standard telephone channels were developed and assimilated by industry in parallel. It is natural that in the future rural telephone networks (STS) built on the basis of the ATS KE and also electronic switching centers, the latter will interact over common signal channels (OKS), the advantage of which is speed of connection, the possibility of almost unlimited expansion of the composition of the telephone signals, and so on. The use of OKS and program-controlled ATS will permit optimal control of the STS as a whole.

One of the important elements of operative network control is dynamic control of the call traffic [1]. In [2] a study was made of the method of servicing calls in the network based on estimating the efficiency of bypass routings. In the case of decentralized call control in the network providing for higher reliability and viability of the STS, at the outgoing automatic telephone offices (ATS) in practice it does not appear possible to estimate the efficiency of each bypass routing individually; therefore in [1], a combined method of dynamic traffic control is proposed in which the game method of flow distribution is combined with the method of limiting the number of bypass routings in the presence of overloads (exceeding a defined threshold) of the segments making them up, which can be used also in the STS not having OKS, that is, in the near future.

The appearance of OKS will permit intense exchange of messages between the automatic telephone offices for operative control of call traffic in the STS. Let us consider the call control algorithm in future STS precisely implementing the method proposed in [2]. The essence of the algorithm consists in the fact that for a call from an outgoing ATS over an OKS network hunting messages (POS) will be propagated in the direction of the incoming ATS. Each of the hunting messages will accumulate a total estimate of the traveled path, and a response message (OTS) will be sent in the opposite direction from the incoming ATS after analysis of the incoming messages and selection of the best of them. This response message will initiate setting up the call over the most optimal routing for the criterion used.

It is convenient to begin the discussion for the STS with a connected OKS network. The configuration of the STS at each ATS is reflected by the relief matrix (MR) [3]. Any ATS outside the STS has information corresponding to it which pertains to the central office (TsS), for the STS is connected through it to the other networks of the country.

Let a request that a call be set up with a subscriber connected to ATS-B arise at ATS-A. If ATS-B is opposing and available for ATS-A over the channel group connecting them, then after exchange of the corresponding messages between them (with availability of the called subscriber), the call is set up. If the ATS-B is not opposing or, being opposing, it is inaccessible along a direct routing, then from ATS-A the POS is transferred to all accessible opposing ATS. In addition to the number information, each POS contains the following: a pointer of the maximum number of tandem sections (MChT) between the ATS-A and the ATS-B (for the TsS, if the ATS-B is located outside the STS), the value of which is taken from the MR; the index of the number of traversed tandem sections (PChT); the total estimate of the traveled routing (SOP) equal to the sum of the estimates of individual tandem sections [2]. The POS transmission sequence to the opposing ATS is determined by the values of the MR numbers located in the row corresponding to the ATS-B number: namely, initially the POS which correspond to the smallest numbers are transmitted.

The routing for reception of the POS is stored in each opposing ATS, and the ATS-B number is checked against it. When they do not compare, the tandem POS transmission program is run: by the ATS-B number from the MR, the corresponding row is calculated, each of the numbers of which is then added to the PChT pointer, and the results are compared with the MChT pointer; thus, the POS are discovered which follow along direct routings (in the general case there can be several of them which are identical with respect to the number of tandem connections). On the accessible routings corresponding to the direct paths, unconditional transmission of the POS is realized (with corresponding corrections of the PChT pointer and values of the SOP). The transmission of the POS over all remaining accessible routings (the reception routing is not considered) is realized under the condition that the POS transmitted previously over them (each routing has a transmitted POS memory) had larger values of the SOP and also under the condition that the values of the SOP prepared for transmission of the POS are less than one (otherwise, according to [2], the bypass routings corresponding to them are not efficient). The POS transmission sequence from the tandem ATS is defined just as for the outgoing ATS.

On comparison of the ATS-B number in the received POS with the number of the ATS receiving it, the POS analysis program for the incoming ATS is run (on availability of the called subscriber). For this purpose, at the ATS-B the received POS having the same number information are recorded for a defined time. Then the POS are selected from among them (if they exist), for which the PChT and MChT pointers are identical (that is, received over direct routings), and the one is selected from them which has the least value of the SOP, although it can exceed one. These POS must arrive at the ATS-B first, which is ensured by the transmission sequence at the outgoing and tandem ATS. In the absence of POS received over direct routings (as a result of blockings on these routings), a choice is made among the POS arriving over the bypass routings having the least value of the SOP (if there are several, then the one having the smallest value of the PChT pointer).

On the reception routing selected as a result of analysis of the POS, the free channel is taken at the incoming ATS-B, and the OTS containing number information and the PChT pointer (with value decreased by one) and also the busy channel number as identifiers is transmitted. On reception of the OTS in the tandem ATS, the channel is busied on the routing receiving the corresponding POS, the value of the PChT pointer of which coincides with the PChT pointer in the OTS (and if there are several of them, the one having the least value of the SOP), and the OTS is again transmitted (with a decrease in the value of the PChT pointer by one and a new channel number), and setting up the call between the selected channels is initiated. On reception of the OTS in the ATS-A, a loop check of the channel is made from it, after which the calling and called subscriber are connected to the channels in the outgoing and incoming ATS. If the OTS does not reach the outgoing ATS-A during the check time, the call is considered lost. Information about the POS is erased in all of the ATS on expiration of some time interval after the beginning of recording of it.

If the incoming ATS-B is located outside the STS containing the ATS-A, then the POS is received at the TsS, which in the given case is tandem, where the availability of the routing to the ATS-B is checked. If it is available, then the OTS is sent in the direction of the outgoing ATS-A, and the call is set up between them; the exchange of control information between the TsS and the opposing ATS of the other network is organized in parallel. Finally, the success of the connection will depend on the availability of the incoming ATS-B and the called subscriber from the TsS. If the outgoing ATS-A is located outside the STS zone, then the call is initially set up from it to the TsS, and then the TsS organizes the hunt for the optimal routing to the incoming ATS-B.

If the STS has a quasi-connected OKS network, then for transmission to opposing ATS with which there is no direct signal channel or information about the MR, POS and OTS vectors, the latter are equipped with a special title and number of the corresponding opposing ATS; this title and number permit the vectors to be distinguished from ordinary messages and to be sent to the indicated addressee via other ATS. For the rest, the transmitted information content and the algorithms for processing it remain the same as in the connected OKS network.

On the level of the terminal offices (OS) in the STS using the OKS for signaling, any of the existing ATS can be connected through the ATS KE. The call control in the network from these ATS is organized from the ATS KE after reception of the number information in them.

Comparatively small size of the STS and high speed of control information processing in the ATS and transmission of it over the signal channels ensure rapid convergence of the search process. Thus, the given algorithm based on almost "instantaneous" state of the network at the time of hunting implements the optimal method [2] of servicing calls in the STS sufficiently precisely.

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SOME RESULTS OF COMPARING TWO METHODS OF CALCULATING LOSSES IN COMMUNICATIONS NETWORKS

[Article by A. I. Movshovich, Leningrad , pp 117-121]

[Text] The accuracy of two methods of calculating losses π_{ij} with respect to communication routings in a channel-switched network in which one-parametric and two-parametric models of traffic description are used, is compared. In method 1, all of the flows in the network are considered Poisson, that is, they are characterized by their own first moments -- average values. For calculation of π_{ij} in this case it is sufficient to have a call traffic distribution plan, and the values B_ℓ of the blocking probabilities of the network branches ($\ell = \overline{1, M}$). In the two-parametric model for each flow two moments are calculated: the average a' and dispersion v'. Here, in addition to the values of B_ℓ, additional operating parameters of the network branches are used, the composition of which is determined by the adopted calculation algorithm a', v'. The Katz algorithm [3] for calculating losses for a model of flows with two parameters is used in method 2.

The accuracy of the analytical methods of calculating losses is determined by comparing the calculated values of π_{ij} with the estimates by calls $\hat{\pi}_{ij}$ of these values obtained using simulation. The length of the random process during the simulation is selected so that the confidence intervals for the estimated values will be appreciably less than the error in the calculation method. The values of B_ℓ in the analytical methods are found as a result of performing the iteration process, in each step of which the incoming load is distributed with respect to the routing trees of all of the communication routings. The error in calculating the probabilities B_ℓ depends on the model of description of the flows and influences the accuracy of the final result.

The magnitude of the systematic error in the loss calculation methods which is determined when calculating π_{ij} by exact values of B_ℓ and other operating parameters of the network branches is of interest. The systematic error is the lower bound of the error in the analytical method and characterizes the accuracy of the selected model of the flows and the methods of calculating the parameters of this model. For calculation of the systematic error in the analytical methods it is possible to use combined methods (KM) of calculating losses in the communications networks [1].

Table 1

	I	2	3	4	5	6	7	8
I	0	24	30	4I	47	52	24	4I
2	30	0	26	35	40	44	2I	35
3	3I	2I	0	35	40	44	2I	35
4	4I	27	34	0	54	60	27	47
5	46	3I	39	53	0	68	3I	53
6	53	35	44	62	70	0	35	62
7	30	2I	26	35	40	44	0	35
8	4I	27	34	47	54	60	27	0

Table 2

	I	2	3	4	5	6	7	8
I	0	I4,I7	I8,9	28,37	33,09	37,8	I4,I7	28,37
2	I9,2	0	I5,37	23,I	26,9	30,75	II,55	23,I
3	I9,37	II,62	0	23,26	27,08	30,9I	II,62	23,26
4	27,9	I6,68	22,33	0	38,99	I4,66	I6,68	33,5
5	32,5	I9,54	25,9	38,6	0	5I,8I	I9,54	38,63
6	38,22	22,88	30,58	45,76	53,3	0	22,88	45,76
7	I9,2	II,55	I5,38	28,I	29,3	30,75	0	23,I
8	27,94	I6,68	22,33	33,5	38,99	44,66	I6,68	0

Table 3

Q	B^M	$\hat{\pi}^M$	δ^M	π_1^K	π_2^K	$\Delta\pi_1$	$\Delta\pi_2$	η_1	η_2
I,I	0,0348	0,0047	±0,0002	0,0024	0,005I	0,0026	0,00II	50,6	2I,4
I,I5	0,0704	0,0I54	±0,0003	0,0098	0,0I80	0,0058	0,0032	34,3	20,4
I,3	0,2306	0,0964	±0,0009	0,0805	0,II05	0,0I60	0,0I76	I6,0	I8,6

Results are presented from calculation of the losses with respect to the KM in two communications networks. All of the calculations are performed using a set of programs for the "Minsk-32" computer described in [2]. In each version the estimates of $\hat{\pi}_{ij}$ were determined by a run with a length of 1.5 million events of the "call" and "release" type, after which the losses π_{ij} were calculated by each of the two methods. The call traffic distribution plan in the two networks was formed by the "relief" method, where among the routes of equal length, preference was given to the route passing through the adjacent junction with smaller number. The relay-race algorithm for setting up calls was used everywhere.

The first network is an example of the district network of a GTS [city telephone network]. The matrices of the capacities of the branches and extensions (in Erl) are presented in Tables 1 and 2. The average busy time is 85 seconds. The channel bunches are unidirectional. Bypass routings two units long are permissible. The network was designed in the region of small, medium and large losses. The extension matrices were calculated for all versions by multiplying the elements of Table 2 by the coefficient Q. The results of the calculations for three values of Q are presented in Table 3. The values calculated for the model with one parameter have a lower index 1, and for the model with two parameters, the index 2. For each version the following are presented: B^M -- the average probability of blocking of the branches with respect to the network; π^M -- the estimate of the average losses over the network with respect to calls; δ^M -- the deviation of the confidence limits with respect to $\hat{\pi}^M$ for a confidence level of 0.95; π^K -- the average losses over the network calculated by the KM; $\Delta\pi$ -- the average systematic error in the analytical method; η -- the relative average error in calculating the lost load with respect to service routings. The values of $\Delta\pi$ and η are determined by the formulas:

$$\Delta\pi = \frac{1}{L}\sum_{i,j}|\pi_{ij} - \hat{\pi}_{ij}|,$$
$$\eta = \left(\frac{1}{Y\hat{\pi}}\sum_{i,j} y_{ij}\,|\pi_{ij} - \hat{\pi}_{ij}|\right)100\%.$$

Summation is carried out with respect to all pairs (i, j), for which $y_{ij} \neq 0$. Here y_{ij} is the load going to the service routing (i, j); Y is the total load imposed on the network; L is the number of service routings.

As a second example, a network was selected, the capacity matrices of the branches and extensions of which are presented in Tables 4 and 5. The channel groups are bidirectional. Calls no more than five branches long are permitted. The average busy time is 5 minutes. In contrast to the preceding example, the network branches primarily service a tandem load. In addition, the network is characterized by small connectedness and branched "trees" of the routes on all service routings. The network was calculated for four values of the coefficient Q. For each version, in addition to the average losses with respect to the network, the average losses with respect to three groups of service routings were calculated. One group included the service routings with identical minimum distance R among the branches between the outgoing and incoming junctions. The results of the calculations are presented in Table 6. For each Q, the average losses with respect to the network are presented in the first row, and the average losses for the first, second and third groups are presented in the three subsequent rows, respectively.

Table 4

	I	2	3	4	5	6	7	8
I	0	20	0	0	0	0	32	I3
2	20	0	43	0	0	0	0	I3
3	0	43	0	25	0	36	0	0
4	0	0	25	0	23	0	0	I6
5	0	0	0	23	0	6	I3	0
6	0	0	36	0	6	0	20	0
7	32	0	0	0	I3	20	0	0
8	I3	I3	0	I6	0	0	0	0

Table 5

	I	2	3	4	5	6	7	8
I	0	2	2	2	0	2	6	2
2	2	0	6	2	2	2	2	2
3	2	6	0	2	2	6	2	2
4	2	2	2	0	2	2	2	2
5	0	2	2	2	0	2	2	2
6	2	2	6	2	2	0	2	2
7	6	2	2	2	2	2	0	0
8	2	2	2	2	2	2	0	0

The results obtained permit formulation of the following conclusions.

1. The single-parametric model of flows leads to low results everywhere. The relative mean error in calculating π_{ij} decreases with an increase in the losses in the network and is satisfactory only in the large-loss region.

2. In the region of small and medium losses the accuracy of method 2 is appreciably higher than the accuracy of method 1.

3. In the region of large losses, the systematic error in the investigated methods is approximately the same.

Table 6

Q	$B^м$		$\hat{\pi}^м$	$\sigma^м$	$\pi_1^к$	$\pi_2^к$	$\Delta\pi_1$	$\Delta\pi_2$	ε_1	ε_2
0,9	0,0550		0,009I	±0,0002	0,003I	0,0064	0,0072	0,0034	66,2	3I,9
		R = 1	0,0047	±0,0002	0,0009	0,0027	0,005I	0,0027	8I,0	42,4
		R = 2	0,0I27	±0,0004	0,0049	0,0096	0,0078	0,0036	62,I	28,3
		R = 3	0,0265	±0,00I5	0,0I24	0,0I97	0,0I4I	0,0075	55,I	27,3
I,0	0,II58		0,0354	±0,0005	0,0I63	0,03I6	0,0222	0,0056	54,2	I3,4
		R = 1	0,022I	±0,0005	0,007I	0,0I79	0,0I9I	0,0060	68,2	I9,5
		R = 2	0,0475	±0,0008	0,0244	0,045I	0,0232	0,0049	49,I	I0,3
		R = 3	0,0824	±0,0026	0,0504	0,0747	0,0320	0,0077	38,8	9,4
I,I	0,I905		0,0834	±0,0006	0,05I2	0,09II	0,0360	0,0098	38,8	II,0
		R = 1	0,057I	±0,0007	0,0283	0,0605	0,0353	0,0049	50,8	9,6
		R = 2	0,I08I	±0,00I2	0,0728	0,I238	0,0354	0,0I58	33,0	I4,6
		R = 3	0,I7I2	±0,0037	0,I275	0,I709	0,0437	0,0029	24,8	I,7
I,3	0,3I5I		0,I964	±0,00I2	0,I6I7	0,2I74	0,0372	0,0266	I7,6	I2,9
		R = 1	0,I45I	±0,00I4	0,I097	0,I6I5	0,04I5	0,0I93	24,4	I3,4
		R = 2	0,248I	–0,0022	0,2I50	0,28I9	0,0333	0,0352	I3,4	I4,2
		R = 3	0,3448	±0,0063	0,3097	0,3329	0,035I	0,0I80	I0,2	5,2

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SET OF PROGRAMS FOR CALCULATING LOSSES IN COMMUNICATIONS NETWORKS BY THE COMBINED METHOD

[Article by A. I. Movshovich and M. Yu. Khokhlova, Leningrad , pp 122-127]

[Text] Basic information is presented on the set of programs (KP) for the "Minsk-32" computer realizing the combined method (KM) of calculating losses in switched communications networks [1]. The KP is a library of program modules, from which operating programs can be assembled for investigating networks with dynamic call traffic distribution control and with different network algorithms for setting up calls.

The programs of the first phase of the KM which perform the simulation are a significantly revised and expanded version of the KP described in [2]. The programs are divided into the following basic groups: initial data input, the formation of a representation of the network structure and the incoming load to the network, calculation of thresholds for tandem calls, setting up and terminating calls dynamic call traffic distribution control, processing and documentation of simulation results. Different simulation models of communications networks are formed by assembling the corresponding control programs in which the composition and order of execution of the modules are defined and the lengths of the common regions of ready-access memory are also given. The common regions contain standard representation of the network structure and other information files depicting the state and results of operation of the communications network for all programs. As a rule, each data file, the length of which depends on the initial data, corresponds to one named common region. When assembling the working program, the length of this region is taken equal to the maximum of the lengths indicated for it in all of the joined programs. Therefore, when reserving in the KP modules only one word of memory in each common region, the possibility of controlling the memory distribution in the model using the control programs is ensured.

The programs in the set executing the second and third phases of the KM consist of programs for calculating the operating characteristics of the network branches, the programs for calculating the losses π_{ij} with respect to service routings and the programs for documenting the results of calculation and simulation. In addition, for this group of modules there are control programs which give the operating sequence of the modules and reserve the ready-access memory regions for additional data files used in the second and third phases of the KM.

The configuration of the communications network is described by the adjacency matrix of the network junctions $[b_{ij}]$ and is depicted in the computer memory by the KNF file. The length of the KNF file is equal to the number of units in the matrix $[b_{ij}]$ and is denoted by M. The KNF elements are ordered in rows $[b_{ij}]$ in increasing order of the numbers j of the incoming junctions. Here each $b_{ij} = 1$ corresponds to one element of the KNF file. Each oriented branch is represented in the KNF by one element, and unoriented branch by two elements with the indices ℓ and $\bar{\ell}$, for $b_{ij} = b_{ij} = 1$ for it. The KNF element (ℓ) takes one word of memory and contains the pointer γ for blocking the channel group, the number $\underline{i}$ of the outgoing junction, the number j of the incoming junction and the index $\bar{\ell}$ of the paired element corresponding to b_{ji}. For oriented branch we set $\bar{\ell} = 0$. Then the index $\ell = \ell(i, j)$ will be called the network number of the branch (i, j), and the junction number of the branch will be defined as $\delta_{ij} = \ell(i, j) - \ell(i, j_1) + 1$ where $\ell(i, j_1)$ is the index of the first branch in the KNF file going out from the junction i.

The junction numbers of the branches are stored in the matrix $[\delta_{ij}]$, $i, j = \overline{1, N}$, the elements of which take up one byte of memory each. For the pairs (i, j) such that $b_{ij} = 0$, the element $\delta_{ij} = 0$.

In the simulation step, for each branch of the network the following parameters are stored: c -- the capacity of the branch, c_n -- the magnitude of the threshold for tandem calls, σ -- the indicator of damage to all channels, ρ -- the branch length, m -- the number of busy channels, $\tilde{a}$ -- the serviced load, t_c -- the total time the group is in the state m = c, t_n -- the total time the group is in the consolidated state $m \geq c_n$. These parameters occupy 8 M words of memory.

In the second and third phases of the KM, four additional information files M words long each are used: the variation coefficients θ, the load imposed on the branch; the variation coefficients θ^n of excess load; the empirical coefficients k; the loads $\tilde{a}^o$ corresponding to the calls connected through the branch but rejected.

The load imposed on the communications network is given by the extension matrix $[y_{ij}]$, where y_{ij} is the average value of the outgoing load from the junction i to the junction j. When constructing the network model, the following assumptions were made: the flow of calls on any service routing is Poisson, the talk time is distributed exponentially, the time for setting up calls and disconnecting the setup calls is equal to zero, there are no repeated calls, the busy time and failure of the subscribers to answer are not taken into account, losses as a result of busy switching and control equipment at the offices are nonexistent. These assumptions permit replacement of the true process of servicing calls by a modified Markov process [3]. After this substitution the KP simulates an embedded Markov chain over a set of calls and releases.

In order to accelerate the operation of the call distribution algorithm with respect to service routings, the set of gravitations $\underline{y_{ij}} \neq 0$ is converted to the vector MT so that $(MT(k + 1) - MT(k))Y = y_{ij}$, $k = \overline{1, L}$. Here Y is the total load imposed on the network, L is the number of gravitations $y_{ij} \neq 0$. Let us note that $MT(1) = 0$, and $MT(L + 1) = 1$. Each MT(k) is placed in correspondence to a pair (i, j) which is stored in the gravitation referencing list (SPT). The pairs (i, j) are placed in the SPT in the order of the nonzero elements according to rows of the matrix $[y_{ij}]$. Thus, for determination of the service routing (i, j), in which it is necessary to send the incoming call, it is sufficient to find the k such that $MT(k) < \xi \leq MT(k + 1)$, where ξ is the value of the random variable uniformly distributed in the interval (0, 1).

The calls that are set up are represented in the format $(i, \delta^1, \ldots, \delta^2)$, where $\delta^1, \ldots, \delta^2$ is the sequence of junction numbers of the branches beginning with junction i. For $r \leq 4$ the call takes one word of memory, and for $5 \leq r \leq 9$, two words of memory. Short calls are recorded as they arrive from the beginning of the SUS file, and long ones, from the end of the SUS file containing both lists of set up calls. On occurrence of the "release" event, the call which must be removed from the SUS file is selected equiprobably among all of the setup calls and the last element of the corresponding list is recorded at the location of the removed element.

The process of setting up calls in the communications networks is simulated by the PROKD program. This program permits investigation of various algorithms for selecting the outgoing routings (VIN) at the network junctions for two methods of setting up calls: relay-race and rescanning (with return to the outgoing office). The outgoing routings are selected at each junction s of the network using the routing matrix $R_s = [h^s_{jk}]$. The matrix R_s has a number of rows equal to the number of network junctions, and a number of columns equal to the number of branches emanating from the junction s. The matrix columns are attached to the outgoing routings in increasing order of the numbers of the adjacent junctions, that is, $k = \delta$. The matrix element h^s_{jk} is a floating-point number, which is an estimate of the set of routings from junction s to the destination j through a branch with junction number $\delta = k$. The initial filling and variation of the matrices R_s during operation of the network are realized by the flow transmission routing control programs. All of the matrices R_s are in a common region for which N x M words of ready-access memory are reserved.

Let us consider the operation of the VIN algorithm at the junction s when setting up a call over a service routing (i, j). The problem of the VIN algorithm consists in calculating the junction number δ of the next branch which is included at the junction s in the established connection. If $\delta = 0$, this means that there are no available routings to the junction j from the junction s. For calculation of δ, the following series of operations are executed.

Step 1. If $\delta_{sj} \neq 0$, the possibility of setting up the call with respect to the branch to the destination j is checked. With positive outcome, we set $\delta = \delta_{sj}$, and the operation of the algorithm ends; otherwise we proceed to step 2.

Step 2. If $r_s = r_M - 1$, $\delta = 0$, and the operation of the algorithm ends; otherwise we proceed to step 3 (here r_M is the maximum admissible number of branches in the connection, r_s is the dynamic length of the connection).

Step 3. The list D of junction numbers of available routings is compiled. For the outgoing calls (s = i) the routing with network number ℓ is considered available if $\sigma(\ell) = 0$, $m(\ell) < c_n(\ell)$, and a tandem connection is permitted through the adjacent junction. For tandem calls, the routing to the preceding junction is also excluded from investigation. In the D list formation cycle the value of δ_* is calculated corresponding to the minimum element h_* in the row j of the matrix R_s among the columns $k \in D$. When hunting for the minimum and filling the list D the row j of the matrix R_s is examined from left to right.

Step 4. If the list D is empty, then $\delta = 0$ and operation of the algorithm ends; otherwise we proceed to step 5.

Step 5. If $\varepsilon_h = 0$, then $\delta = \delta_*$, and operation of the algorithm ends; otherwise we proceed to step 6.

Step 6. The list D_e of junction numbers of equivalent outgoing routings is compiled. It includes δ_* and all $\delta \in D$, for which $|h^s_{j\delta} - h_*| < \varepsilon_h$. The desired value of δ is selected from the list D_e equiprobably, and the operation of the algorithm ends.

The set of above-investigated network algorithms for setting up calls can be described as a function of three parameters $A(w, r_M, \varepsilon_h)$. Here w is the maximum number of attempts to reestablish connection. The value of $w = 0$ corresponds to the relay-race algorithm of setting up calls. The value of ε_h determines the equivalence level of estimates of outgoing routings in the routing matrices. Varying ε_h, it is possible to change the degree of influence of the control algorithms U on the call traffic distribution in the communication network.

In the KP, two control systems are realized for the flow transmission routings: a combined system based on the relief method and the relief method with inertia $U_p(n, T)$ [4] and a combined system based on the relief method and the game method $U_H(\alpha, \beta)$ [5]. Here n is the inertia parameter, T is the minimum vector exchange period, α is the magnitude of the penalty, β is the magnitude of the reward. In order to standardize the VIN algorithms, in contrast to [5], it was assumed that $\alpha > 1$, $\beta < 1$.

Both of the combined control systems presuppose the presence of a relief matrix $H_s = [H^s_{ij}]$ at each junction s of the network. This matrix has the same structure as the matrix R_s, but the elements H_s occupy one byte of memory each [4]. The structural control is realized by network orientation programs which calculate the finally reformed network relief $\{H_s\}$, $s = \overline{1, N}$. On the basis of the obtained relief, the matrices R_s are corrected, after which the load control is included: the relief method with inertia or the game method.

When realizing the method $U_p(n, T)$, modification of the relief method proposed in [6, 7] is generalized. The refinement pertains to the blocking rule for the columns of the matrix H_s when calculating the minimum vectors (MV). Now the algorithm for calculating the minimum in row j of matrix H_s coincides with the algorithm for calculating h_* in row j of matrix R_s (step 3 of the VIN algorithm) for a tandem call coming from the junction to which the MV will be sent. After calculating the MV, its components corresponding to adjacent junctions with forbidden tandem connection are set equal to one if $m(\ell) < c_n(\ell)$. During operation of the programs in the structural control mode, the MV calculation algorithm coincides with the investigated one if we consider that $m(\ell) = 0$ for $\ell = \overline{1, M}$.

The programs of the second phase of KM calculate the operating parameters of the network branches. By the magnitudes of the service load $\tilde{a}_\ell$ and probability B_ℓ of blocking the branch, the parameters θ_ℓ, θ_ℓ^n and K_ℓ are calculated for $\ell = \overline{1, M}$. The calculation algorithm is discussed in [1]. When giving a single-parametric traffic model we set $\theta_\ell = 1$, $\theta_\ell^n = 1$, $K_\ell = 0$, where $\ell = \overline{1, M}$.

The programs of the third phase of the Km calculate the losses π_{ij} with respect to the communications network routings (i, j). The algorithm for calculating π_{ij} was developed for the network algorithms for setting up calls of the $A(w, r_M, 0)$ type. Another restriction is the assumption that $c_n(\ell) = c(\ell)$, $\ell = \overline{1, M}$.

The load distribution y_{ij} with respect to the routing "tree" from junction i to junction j is realized by the RTYaG program. The routing tree in this case is not formed in advance, but each new routing to junction j is constructed as needed by the MARSh program. This program uses the vector NPV which is a function of the number j of the incoming junction. The NPV vector contains a sequence of network numbers of the branches ℓ_s^1, ℓ_s^2, ..., ℓ_s^n for each junction s of the network. These branch numbers are arranged in increasing order of the numbers of the paths of selecting the outgoing routings at junction s when setting up a call to junction j. Here ℓ_s^n is the network number of the branch which is the last-choice path. The application of the NPV vector ensures independence of the MARSh program with respect to the VIN algorithm selected in the model.

The general calculation scheme using the RTYaG program consists in the following.

Step 1. The route is constructed from junction i to junction j. The number of the path of choice q for the junctions newly included in the route is equal to one. The load is distributed along the route. Here the excess loads from all branches of the constructed route are stored.

Step 2. The transition is made to the next to the last junction k of the route, and the last junction is excluded from the route. At the junction k, we set $q = q + 1$.

Step 3. The element ℓ_k^q is extracted from the NPV vector. If $\ell_k^q = 0$, transfer to step 4, otherwise transfer to step 5.

Step 4. For the relay-race algorithm of setting up calls, the excess load from the network ℓ_k^{q-1} is considered lost, and for algorithms with rescanning it is added to the excess load from the branch ℓ_i^q. If k = i, transfer to step 7; otherwise transfer to step 2.

Step 5. The route is constructed from junction k along the path of choice with number q in the direction of the junction j.

Step 6. The excess load from the branch ℓ_k^{q-1} is distributed by the constructed route, transfer to step 2.

Step 7. The value of π_{ij} is calculated.

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APPLICATION OF SPECIALIZED PROCESSORS IN THE CONTROL UNITS OF SWITCHING CENTERS

[Article by A. G. Popova, Moscow , pp 127-130]

[Text] At this time the opinion exists that flexibility characteristic of the programmed approach and connected with using finished standard components (microprocessor sets) permits the advantages of programmed control in the switching centers to be increased even more. The central control unit of the switching center (KU) can be constructed as a central processor (TsP). Here two methods of executing the operating algorithm of the control unit are distinguished: hardware and microprogram.

In the hardware implementation of the algorithm the central control unit is a programmable device which consists of logical elements and memory elements joined together in a defined way. The disadvantage of such programmable devices is irregular structure, but the speed of these devices is high. In the control hardware, the sequence of execution of the program instructions is determined by the rigid connections between logical elements built into the system, that is, rigid interelement couplings.

For the microprogram realization, the central control module consists of an information processing module (BOI) and microprogram device (MPU). The BOI performs functions analogous to the functions of the arithmetic-logical circuitry of a computer except that in the KU the majority of functions are logical. The MPU realizes the microprogram execution of the instructions of the operating algorithm of the KU control unit.

One of the possibilities for increasing the operating efficiency of the control units (UU) is the application of sufficiently autonomous devices having their own control. These devices are called specialized processors (SP). The application of specialized processors is expedient in cases where the operating algorithm of the processor is frequently used in the common operating program of the UU, and the operating program of the processor is distinguished by sufficient simplicity and has little connection to the remaining operating programs of the UU.

The peripheral control units (identifiers, modules for including the switching elements, and so on) [1] can be constructed as SP. The construction of the KU control units in the form of central and specialized processors permits use of the advantages of both the centralized and distributed methods of control in the KU.

One of the basic missions of the KU control unit is the search for a connecting path in the switching system. The initial information for beginning the search for a connecting path is the input and output numbers between which it is necessary to set up the call and the final result is the values of the connecting path coordinates selected for connection or the signal of impossibility of setting up the call.

When carrying out the mission by a specialized processor, the latter can contain MPU, which ensures execution of the hunting program instructions, BOI and OZU in which the data are stored regarding the state of the connecting paths and the set up calls, the SP operating program and auxiliary values required to execute the program. The SP operating program will contain the following stages: processing of requests from the TsP, the search for a connecting route, recording of information about the set up call, release of the connecting path after ring-off, organization of central processor requests. Finding the connecting path includes successive search for a free switchboard output in each of the elements. The process of finding the free switchboard output on a link of an i,z-link KS consists in the following. After determining the address, the ready-access memory cell and reading the required bits of the word in this cell, a number will be entered in the m_i last bits of the operating cell R_i' each bit of which characterizes the state of one output of the required switchboard of link i. In order to decrease the nonproductive busy time of the specialized processor in the case of absence of free outputs in the given switchboard it is necessary to introduce the operation of checking for the presence of at least one unit in the number obtained. If there is at least one unit, then the search for this unit in the number is made, that is, the number of the bit is determined in which the unit is entered. If there is no free output, then a return is made to the search for another free switchboard output of the preceding link (i - 1).

When checking the state of the switchboard output, logical bit-by-bit multiplication of the register contents R_i' and the contents of the auxiliary register R_i''' is carried out, in one bit of which a one is entered and in the rest, a zero. In the initial position before the beginning of the search for the connecting path a one is entered in the low-order bit of the register R_i'''. If a number having a zero in all bits is obtained as a result of multiplication, this means that the checked output is busy. Therefore it is necessary to increase the number of the checked output by one. Simultaneously, a shift of the register contents R_i''' by one bit to the left is made, and the state of the next switchboard output is again checked.

The formation of the ready-access memory cell address for selecting information about the state of the outputs of the accessible switchboard on link (i + 1) consists in the following. On link i the intermediate line is determined by the set of coordinates α, β, γ, ..., δ, the values of which are entered in the cell R_i^{iv}. Here the coordinate δ corresponds to the number of the switchboard output selected as a result of performing the preceding operations. In the next link (i + 1) the intermediate line is defined by the set of coordinates β, γ, ..., δ, ρ, and the value of the last coordinate ρ will be defined when finding the connecting path on link (i + 1).

Thus, for determination of the address of the cell corresponding to the intermediate lines of the next link, it is necessary to enter the register contents R_i'' (the code of the selected switchboard output of link i) in the low-order bits of the selected output code of link i; then, shifting the register contents R_i^{iv} by $\log_2\alpha$ bits to the left, the obtained number is used to determine the address of the ready-access memory cell.

This method of forming the memory cell address is suitable for the KS with any values of the structural parameters under the condition of entering the code of the KS target in the binary-decimal number system. If the structural parameters of the KS are defined as

$m_i = 2^k, n_i = 2^\ell,$

then the word length of the code of the object is made best use of, and the shortest word-length of the MPU registers of the specialized connecting path finding processor is required.

The proposed SP operating algorithm for finding the connecting path can be used for the KS of any structures and with arbitrary number of links.

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SYSTEMS APPROACH TO CONTROL SYSTEMS AT THE JUNCTIONS WITH DIFFERENT METHODS OF DELIVERING MESSAGES TO THE NETWORKS

[Article by V. N. Roginskiy, Moscow , pp 130-132]

[Text] The application of different methods of delivering messages in channel-switched (KK), message-switched (KS), packet-switched (KP) and other networks was until recently connected with different forms of service and different technical implementation. For comparison, different approaches and methods of estimation were used. All of this leads to the fact that it is impossible to make an objective comparison or estimate the advantages or deficiencies of each procedure. An effort is made to evaluate various methods of delivery and the requirement on control of the switching centers from united points of view.

All of the methods of delivering messages in the network can be divided into two groups: without switching centers (with direct or common channels) and with switching centers (UK).

The second group of networks is of the greatest interest. In the general case it is possible to have a combination of different methods in individual sections or for different types of information in the network.

In networks with UK it is possible to isolate two methods of distributing messages in the centers: without accumulation of messages at the centers and with accumulation of messages (or parts of them -- packets).

In junctions without accumulation for each message the routing is selected by address, and either a through transmission channel from the coupled-out channels is set up for each message (channel-switched system) or "armoring" of the channels takes place along the entire path so that on arrival of the next segment of the message it will be transmitted without delay (address-code switched, statistical multiplexing, some of the KP systems -- creation of "virtual," "logical" channels, and so on). If there is no free channel at the given time for the given message, it either waits at the junction or a rejection occurs (the call is lost), which corresponds to waiting at the terminal point (OP). In both cases there is a delay in assigning a channel for transmission of the message.

At centers with accumulation, each message (or packet) is received and stored together with the address, and only then is it transmitted farther as a free channel is assigned (KS and certain KP systems). When the channels are busy, the message waits. Under other equal conditions, the waiting time will be the same as for KK for the same load. On transmission by packets, the waiting time for individual packets will be less, but it can increase for the entire message.

If we consider that there are no losses of messages (information) (the subscribers are "absolutely persistent"), for identical transmission and processing rates and message sizes the busy condition of the channels in any case creates a delay. The message is delayed either at the OP (in the special case, at the man) or at the UK which requires corresponding memory at the switching center. The greatest delays are created when the called OP is busy.

In all cases the control unit (UU) of the center receives and analyzes the address and determines the outgoing routing, finds and assigns a free, unblocked channel and gives the instructions to transmit the address and messages farther, and if necessary it assigns a channel for putting the message in a queue or rejecting it.

In addition, the UU performs a number of other functions: effecting priorities, finding alternative routings, circular transmission, and so on. The realization of these functions is different for different switching methods.

The basic quality index -- the message transmission time in the network for different methods of delivery and identical transmission and processing speeds -- contains a common component including the following times for passage through k junctions: the address transmission and waiting (k times) and single message transmission. For different delivery methods the following are added:

for KK, the waiting time connected with an increase in load on the first channels on the path;

for KS, the time of (k - 1)-fold transmission of the message;

for KP, the time of (k - 1)-fold transmission of one packet and (h - 1)-fold transmission of the address (h is the number of packets in the message).

The optimalness of delivery is determined by the relations of the message and network parameters, admissible delays, the required memory size and other indices when considering additional requirements such as response speed or the requirement of dialogue. In the latter cases, it is necessary to use duplex channels or "armoring" of the channels in both directions.

The greatest effect from statistical multiplexing is contained in cases where the message contains interrupts in information transmission. All of this must be considered when comparing different methods of transmission of messages in the network.

The united approach to estimates of different methods of delivering messages in the network and methods of implementing them will permit standardization and integration of the channel-forming, switching and control equipment, optimal construction of the networks on a united technological base with united control programs and algorithms, selection of the solution for individual parts so as to ensure optimalness on the whole.

DYNAMIC CONTROL OF BRANCH CAPACITIES IN A CHANNEL SWITCHING NETWORK

[Article by O. F. Sergeyeva, Moscow , pp 132-135]

[Text] At the present time a great deal of attention is being given to improvement of the efficiency of the use of communications media, which will permit improvement of the quality of servicing in the network with minimum expenditures of financial resources. One of the means of solving this problem is operative correction of the network structure for different operating periods when the dynamic traffic control in the network does not ensure the required servicing quality for certain pairs of offices.

A study is made of one of the problems of operative variation of structure with variation of the situation on the network consisting in correcting the capacities ensuring optimal (from the point of view of efficiency of use of the network channels and service quality) loading between each pair of uniformly gravitating offices in the network structure. The necessity for such correction will occur, for example, in the following cases:

when servicing the load flows with noncoinciding peak loading hours (plh), when at different times of day it is necessary to increase the channel groups on certain routings and decrease them on others;

with an increase in incoming load on individual routes as a result of including new groups of subscribers;

with sharp changes in loading of individual network groups, for example, in case of failure.

In these cases it is necessary to correct the network branch capacities without changing the topology of the network so as to ensure sufficient carrying capacity to service the load flows between each uniformly gravitating pair of offices with given service quality. In contrast to the analogous statement of the problems in [1, 2], in the given paper a study is made of the channel switching network with arbitrary loading of the branches. When solving the investigated problem it is proposed that the following are known:

1. The network structure: number of offices N, number of branches M, their interrelation and capacities of the branches -- $B = \|B_{ij}\|$.

2. Uniform gravitations (load in Erlangs) between pairs of offices -- $F = ||\phi_{ij}||$.

3. The admissible magnitude of the losses for each uniformly gravitating office is $P(i, j)_{ad}$.

4. The admissible magnitude of losses in the branches and the corresponding average use of the lines in the channel groups forming the branches $\eta_{min} - \eta_{max}$, where $\eta = (Y_{const} - Y_{obsl})/n_{channel}$.

5. The load traffic distribution algorithm with respect to network branches.

In the given paper when solving the problem of correcting the branch capacities, optimizing the carrying capacity of the network, the concept of saturated cross section is used [3]. This approach is being widely used abroad at the present time to optimize the network topology under the name of the cross section saturation method. The basic idea of this method consists in the fact that an increase in carrying capacity of the network is achieved as a result of increasing the carrying capacity of the saturated cross section. In the general case the cross section is the minimum set of branches, on removal of which the network becomes unconnected. In the given paper by the network cross section we mean the maximum set of branches on removal of which all of the paths are broken for servicing the load and for some pairs of offices it is impossible to set up a call. Here the cross section is considered saturated if it consists of branches, the use of the lines of which is higher than the maximum allowable amount for the investigated network.

Thus, the presented problem of correcting the network branch capacities to ensure the required carrying capacity of the network reduces to minimizing a function of the type:

$$\min \sum_{\beta_j \in \bar{M}^*} \delta_j (\Delta a_j^{k\ell}), \qquad \begin{array}{l} k = 1, \ldots, N^* \\ \ell = 1, \ldots, N^* \\ j = 1, \ldots, M^*, \end{array}$$

where
$$\delta \begin{cases} > 0, \text{ if } \beta_j \subset \bar{S}_H \\ < 0, \text{ if } \beta_j \not\subset \bar{S}_H, \text{ but } \eta_j < \eta \text{ min}, \\ = 0, \text{ if } \beta_j \not\subset \bar{S}_H, \text{ and } \eta_j > \eta \text{ min}; \end{cases}$$

$\underline{S_H}$ is the set of saturated cross section branches;
$\overline{N}^*$ is the subset of junctions separated by the saturated cross section S_H;
$\overline{M}^*$ is the subset of network branches making up the saturated cross section and also the branches entering into the paths for servicing the load between pairs of offices separated by this cross section S_H;
$\Delta a_j^{k\ell}$ is the increment of the serviced load on the β_j-th branch when servicing the load between the pair of offices (k, ℓ). The following condition must be satisfied here:

$$P_{ij} \le P(i, j)_{ad}, \qquad \begin{array}{l} i = 1, \ldots, N, \\ j = 1, \ldots, M. \end{array}$$

After the values of the total load increments -- Δa_j^{Σ} on the corrected branches are defined, the variations of the loss probabilities on these branches corresponding to them are defined as follows:

$$\Delta p_j = -\frac{\Delta a_j^{\Sigma}}{A_j^{\Sigma}},$$

where A_j^{Σ} is the total load arriving for servicing on the β_j-th branch in accordance with the given distribution algorithm before correction.

The obtained magnitudes of the losses on the branches are used to determine the capacities of the branches ensuring the required quality of servicing for the investigated pairs of offices.

It is obvious that the proposed method is heuristic and is realized by successive approximations to the local optimal solution. In this paper an algorithm is proposed which realizes the given method, permitting ensurance of operative control of the network branch capacities under conditions where the dynamic flow control does not permit the required quality of servicing to be ensured for certain pairs of offices on the network. The proposed algorithm includes three basic procedures: determination of the magnitude of the saturated cross section, determination of the required volume of corrections of the branch capacities ensuring the required quality of servicing; calculation of the network parameters characterizing the servicing quality.

The advantage of the given method by comparison with others [1, 2] lies in the fact that it permits a united approach to the solution of the problem of correcting the branch capacities and topological optimization of the network using the cross section saturation method.

When comparing the proposed branch capacity correction algorithm, for example, with the algorithm of [1] it turned out that:

when using the proposed algorithm the same branches are corrected as when using the algorithm of [1];

determination of the required number of corrections ensuring the required carrying capacity requires a smaller number of calculations by comparison with the algorithm [1].

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MATHEMATICAL MODEL AND ESTIMATING THE EFFICIENCY OF A MESSAGE DELIVERY CONTROL TECHNIQUE BASED ON JOINT APPLICATION OF EXTERNAL AND INTERNAL PRIORITIES

[Article by V. N. Silayev, Yu. F. Zolotykh and L. M. Bondar', Moscow , pp 135-138]

[Text] The use of external priorities (fixed priorities) and internal priorities (dynamic priorities) is widely known [1-2].

The novelty of the method proposed in [4] consists in the application of external and internal priorities to control the delivery of messages in a finite control time $u_i > w_i(t)$, $i = \overline{1, p}$, where $w_i(t)$ is interpreted as the actual time spent by the i-th priority request in the system. The value of the priority function $\beta_i(t)$ is calculated by the Jackson formula [2]:

$$\beta_i(t) = u_i - w_i(t).$$

The operation of the network is described by the following procedure. The directive time of delivery u_i, $i = \overline{1, p}$ is assigned to the messages entering a network of i-th priority class. The messages are placed in a source-junction queue in accordance with the priority function $\beta_i(t)$. The highest priority corresponds to the least values of $\beta_i(t)$. After transmission of the i-th class message to the network it is stored for a time T_i (the waiting time for obtaining verification of proper reception by the destination junction of the i-th class message). If on expiration of the time T_i verification has not been received, the message is again transmitted. Otherwise the next message is transmitted with least value of the priority function $\beta_i(t)$. The repeated messages reach the repeated-message queue and have a relative advantage in servicing over newly arriving messages.

Under the conditions of simplest flows of incoming messages with intensity λ_i, $i = \overline{1, p}$, constant transmission time equal to a, independence of the repeated transmissions, the basic expressions are obtained for the Laplace-Stiltjes transformation of the transmission time distribution function in the network, the average waiting time $\overline{w}_i$ in the source-junction, the average transmission time $E[V_i]$ through the network of the i-th class messages for steady state of the network.

Let $B_i(t)$ be the distribution function of the random variable -- the time interval from completion of message transmission to acknowledgment of receipt in the transmitting junction under the condition that verification is delivered on time.

The probability of the last event will be denoted by π_i; then the probability of repeated transmission is defined as $\nu_i = 1 - \pi_i B_i(T_i)$. For $E[V_i]$ the following expression is valid:

$$E[V_i] = \{a + \nu_i[a + T_i + S'(1)a + \tau] + \pi_i \int_0^{T_i} t\, dB_i(t)\}/(1-\nu_i), \qquad (1)$$

where $S'(1) = [S_0 q_0 (1 - z)/(Q[z] - z)]'_{z=1}$ is the average number of messages in the repeated message queue; $Q(z) = \sum_{k=0}^{p} q_k z^k$ is the generating function of the number of arrivals in the repeated message queue; q_k is the probability of simultaneous arrival of k messages in the repeated message queue; τ is the average repeated transmission waiting time of a message from the repeated messages entering the queue.

From (1), $T_i^* = T_{i\,min}$ is found -- the minimum average transmission repetition time (it is rounded to the nearest multiple of a).

For average waiting time $\overline{W}_i$ the following expression is valid:

$$\overline{W}_i = w_0 + Q'(1)\left[\frac{\overline{w}_i - w_0}{a}\right] a + a\Big(\frac{s_0 q_0 Q''(1)}{2(1-Q'(1))^2} + \sum_{j=1}^{i} \lambda_j \overline{w}_j + \qquad (2)$$

$$+ \sum_{j=i+1}^{p} \lambda_j (\overline{w}_j - u_j + u_i) + \sum_{j=1}^{i-1} \lambda_j (u_i - u_j)\Big),$$

where w_0 is the average time of completion of servicing equal to a/2.

The Laplace-Stiltjes transformation of the distribution function of the actual transmission time of an i-th class message is as follows:

$$f_i^*(s) = \sum_{n=1}^{\infty} \nu_i^{n-1} e^{-sna} e^{-s(n-1)T_i} b_i^*(s)\pi_i \sum_{k=0}^{\infty} e^{-sak} s_k \sum_{m=1}^{p} \frac{1}{m} \sum_{l=1}^{m} e^{-sa(l-1)} \cdot q_m^i, \qquad (3)$$

where $b_i^*(s) = \int_0^{T_i} e^{-st} dB_i(t)$, q_m^i is the probability of arrival of m messages in the repeated message queue, one of which is i-th class.

The value of q_m^i is found from the expression:

$$q_m^i = \sum_{K \in A(m-1,i)} \rho_i \nu_i \rho_{k_1} \nu_{k_1} \cdots \rho_{k_{m-1}} \nu_{k_{m-1}} (1 - \rho_{k_m} \nu_{k_m}) \cdots (1 - \rho_{k_{p-1}} \nu_{k_{p-1}}),$$

where $A(m - 1, i)$ is the set of combinations, the powers $|A(m - 1, i)| = C_{p-1}^{m-1}$, numbers $1, 2, \ldots, n, \ldots, p$; $n \neq i$; $k_j \in K$.

The efficiency of the given control technique was estimated by simulating networks of two structures -- distributed, consisting of 27 junctions, and a linear 5-junction network. The networks with and without control with direct servicing procedure were compared by the following indices: the number of messages delivered on time, the average delivery time and dispersion of the delivery time. Under the conditions of uniform loading of the network with the control consisting of 27 junctions, efficiency was demonstrated with respect to the criterion of the average number of messages delivered on time. For a linear network, the range of loads in which control turned out to be efficient was obtained with respect to each junction. In the indicated load range the effect is 5 percent.

The following features are characteristic of the proposed technique:

a) on satisfaction of the condition $\beta_i(t) \leq 0$ the message is lost;

b) the parameter u_i is not the only parameter for all requests (messages), but it is in one-to-one correspondence with the initial priority, the urgency category of the given request;

c) the priority function $\beta_i(t)$ of the request achieved in the given processing stage (in the given communication center) appears as the final priority in the next stage;

d) for $\beta_i(t) = \tau$ the request is assigned the highest priority with which the i-th class request comes to all subsequent network reserves over its repetition route. The interval τ is interpreted as the network parameter and characterizes the time required for the i-th request to pass through the network in the absence of control and loading of the network on a given level in the quasi-exclusive mode of using the processing reserves of the network;

e) the initial and dynamic priorities are selected from the series: $\tau, \alpha\tau, \ldots, \omega\tau$, where $\alpha, \ldots, \omega$ are integral coefficients. Thus, all of them are multiples of τ, and each lower one is a multiple of the preceding higher one.

Physically the efficiency of the control imposed on the network is based on increasing the use coefficient of the network reserves most loaded at the given point in time by expanding the period of their greatest loading in time as a result of the time reserve $u_i - w_i(t)$ and use of the time reserve $\Delta t = u_i - \tau$.

A study was also made of the laws of variation of priority in accordance with arithmetic or geometric progressions or combination of them and also various versions of organizing the time service. Analysis of the results of using different laws of priority variation demonstrated that in the majority of practical solutions the use of arithmetic or geometric progressions gives satisfactory results.

With programmed realization of the enumerated versions it was considered that the network junction has a separate program performing the functions of priority control and monitoring of the time of existence of the message in the network, and the operations system has the apparatus for starting these programs after time quanta equal to or proportional to τ. The volume of the program execution for different versions turned out to be in practice identical: 25-50 instructions of the M-7000 computer, SM-1, SM-2 computers. The difference in versions was discovered in the frequency of starting the programs and the required size of the field for entering the achieved priority in the message title and the time left before expiration of the control period.

The performed analysis of various versions of program implementation permitted substantiation of the choice of program implementation of the method of controlling message delivery on the switching level of a distributed operations system.

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3

STUDY OF METHODS OF ORGANIZING CALL SERVICING IN THE CONTROL COMPUTER OF A COMPUTERIZED SWITCHING CENTER

[Article by A. V. Solov'yev, Moscow , pp 138-141]

[Text] Efficient organization of call servicing in a control computer (EUM) can be achieved when the operating program dispatching algorithms have the possibility of flexible redistribution of its time reserves. This becomes possible with the introduction of interrupts, the use of static dispatching disciplines, formed by composition of transfer rules from one queue of requests to another and selection of queues (programs) and also with restriction of the EUM loading in case of overloads.

The entire set of operating programs of the EUM $S = \{s_1, \ldots, s_{N_S}\}$ is divided into groups $S_1, \ldots, S_L$ so that $S_\ell \cap S_{\ell'} = \Phi$, $\ell, \ell' = 1, \ldots, L$, $\ell \neq \ell'$. The program $S_j \in S_{\ell,j} = 1, \ldots, N_S$ is characterized by the order number in the group and the group number, that is, $s_j \equiv s(i, \ell)$, $i = 1, \ldots, I_\ell$, $\ell = 1, \ldots, L$. The interaction between different groups of programs is defined using the matrix $H(t) = \|h_{\ell\ell'}(t)\|$, the elements of which $h_{\ell\ell'}(t) = 1$ if any level program ℓ can interrupt the performance of any level program ℓ'. In cases where interruption of the ℓ level program by ℓ' level programs is not permitted, the element $h_{\ell\ell'}(t)$ of the matrix $H(t)$ is equal to zero.

On completion of execution of the program $s(i', \ell)$, the transition to another program $s(i'', \ell)$ is made in accordance with the following rules:

A_1 -- on completion of servicing of each request from the queue;

A_2 -- on detection of the control request in the queue (the control request is placed at the "tail" of the queue before the beginning of servicing of the requests in it);

A_3 -- on elimination of requests in the queue.

The program $s(i'', \ell)$ is selected in accordance with the rules:

$B_i = i'' = \min i,\ i = 1, \ldots, I_\ell$

$B_2 - i'' = \xi_j$, if $i = \xi_{j-1}(i' = \xi_k$, then $i'' = \xi_1)$, ξ_1, ξ_k, ξ_{j-1},

ξ_j are the terms in the sequence $\xi = \xi_1, \ldots, \xi_{j-1}, \xi_j, \ldots, \xi_k$, such that $\xi_j \in \{1, \ldots, I_\ell\}$.

As a result of the composition of the rules $A\nu_1$, $B\nu_2$ ($\nu_1 = 1, 2, 3$, $\nu_2 = 1, 2$) dispatching algorithms are formed which are most frequently used when organizing the call servicing in the EUM.

The average interrupt time of a request for the program $s(i, \ell)$ in the EUM for the investigated model of organizing the call servicing is

$$t_{A\nu_1 B\nu_2}(i,\ell) = \left[w^{A\nu_1 B\nu_2}(i,\ell) + b(i,\ell)\right]\left(1 + \frac{R_\ell}{1-R_\ell}\right),$$

where $w^{A\nu_1 B\nu_2}(i, \ell)$ is the average waiting time for servicing of the requests (i, ℓ) when using rules $A\nu_1$, $B\nu_2$ in the call servicing model without interrupts, $B(i, \ell)$ is the average time for execution of the program $l(i, \ell)$, R_ℓ is the total load factor of the EUM by the programs $s(i, \ell)$ for which $h_{\ell\ell'} = 1$.

Since in a number of cases the EUM cannot provide for processing all of the incoming information in real time, overloads occur. The efficient organization of the servicing of calls in the EUM in the presence of overloads is achieved when using adaptive load limitation algorithms [1]. The algorithm for limiting the load of the EUM which can operate jointly with any algorithm in the multilevel model of organization of call servicing is based on using alternative operating programs and the collective of finite automata functioning in random media.

The program s_i, $i = 1, \ldots, N_A$ is designated as alternate if a request to the EUM can be processed by any of the subroutines $s_i^{\nu_i}$ ($\nu_i = 1, \ldots, N_i^a$). The preference in processing the request by subroutine $s_i^{\nu_i}$ decreases with an increase in ν_i. With an increase in ν_i, the values of the average execution times of subroutines $s_i^{\nu_i}$ decrease. The subroutine $s_i^{\nu_i}$ is selected by the finite automaton Π_i on completion of the operations f_{ν_i} ($\nu_i = i, \ldots, N_i^a$). The automaton operates in the random medium $C(t) = \{C_i^{\nu_i}(t)\}$. The probabilities of penalties $C_i^{\nu_i}(t)$ are ordered as follows:

$$C_i^1(t) \geq C_i^2(t) \geq \ldots \geq C_i^{\nu_i}(t) = C_i^{\nu_i+1}(t) = \ldots = C_i^{N_i^a}(t).$$

The automaton Π_i must select the operation f_{ν_i} with greatest probability during the operating process.

The interaction of the automaton Π_i, $i = 1, \ldots, N_A$ is realized in accordance with the procedure analogous to the "common money drawer" in games of disposition [2]. The automaton Π_i fixes the values of the overload indices χ^{wait} and χ^{flow} obtained at the end of the interval T_k. The average penalty for the player played by the automaton in one interval T_k is calculated by the formula

$$a = (\beta_{wait}\chi^{wait} + \beta_{flow}\chi^{flow})/m(\sigma_n)$$

where β_{wait}, β_{flow} are the coefficients selected as a function of the composition of the alternate programs, $m(\sigma_n)$ is the power of the set of automata Π_i selecting the subroutines $s_i^{\nu_i}$. The automata $\Pi_i t \sigma_n$ are penalized in cases where $a > u_i^{\nu_i}$, where $u_i^{\nu_i}$ is the weight of the program $s_i^{\nu_i}$.

For investigation of the efficiency of the call servicing organization algorithms in the EUM, a simulation model and program corresponding to this model were developed. Two groups of experiments were run. The call servicing organization model for the first group of experiments did not contain the algorithms limiting the load of the EUM under overload conditions. The operating programs were dispatched on two levels: priority (reception and output of information) and basic (information processing) with EUM loading coefficient that varies in the range of 0.94 to 0.98. The second group of experiments was run for the same values of the characteristics of the external environment of the EUM as the first, but with organization of call servicing, an adaptive algorithm was used for limiting the load which was based on the operation of the dispatch automata.

The simulation of the EUM program dispatch algorithms demonstrated that in the operating range of variations of the values of the EUM load coefficient not exceeding 5 percent, situations of local overloads arise. The basic level programs directly connected with the priority level of servicing the calls in the EUM turn out to be especially unstable with respect to the local overload situations.

The adaptive EUM load limitation algorithms permit elimination of local overload situations. The gain achieved here with respect to the values of the admissible times the requests remain in the EUM is within the range of 18-36 percent. The use of the dispatching automata made it possible to redistribute the time reserves of the EUM as a function of the overload.

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SOME METHODS OF IMPROVING THE CARRYING CAPACITY OF A CHANNEL SWITCHING NETWORK

[Article by G. L. Slepova, Moscow , pp 141-144]

[Text] Calls in the channel switching networks (KK) can be serviced not only by the system with rejects, but also by a combined system with rejects and waiting. In order to estimate the operating quality of the system with waiting, the most important indices from the point of view of the network user were investigated: the probability of reject -- p -- and the average waiting time -- $\bar{\tau}$ -- and also the most important index from the point of view of operating efficiency of the network -- its carrying capacity.

One of the reasons for using the combined servicing system (KSO) in the KK networks is the effort to increase the carrying capacity of the network inasmuch as by comparison with systems with rejects, the carrying capacity and use of the service units in the KSO are higher because the serviced load is higher.

The combined servicing system in communications networks is implemented by using limited waiting of the calls for the release of channels. Here restrictions are imposed on the time spent by the call in the system. On the existing networks KSO with a limited number of waiting places -- m -- and with a limited number of waiting places and limited waiting time τ_{max}, have found application.

It is necessary to determine whether there is an increase in carrying capacity of the KK network with bypasses on introduction of the KSO and the degree of this increase. For this purpose a study was made of the nature of variation of losses and average waiting time in the network on variation of the number of waiting places for fixed values of the incoming flow and number of channels in the groups. These relations were obtained by analytical calculation (1) and they were confirmed by the statistical simulation method (2) for a fully connected five-junction symmetric network with a number of channels in the group $n_i = n = 30$ with static information distribution plan. Here it was determined that for small incoming loads (to a specific incoming load of $\kappa = 0.8$ Erlang), the introduction of waiting places on the service routings (channel groups) and increasing the number of them lead to a significant decrease in the losses for insignificant values of the average waiting time comparable to the hardware component of the call setup time in the network. With an increase in incoming load above the indicated magnitude, introduction of the KSO is not efficient, for with an increase in number of waiting places both the losses and the average waiting time increase. This increase in losses is possible as a result of nonproductive busy time of the channels in the waiting time, for with an increase in load, the number of waiting calls and their waiting time increase sharply. Thus, the introduction of KSO on the KK networks without using special control methods is efficient only for low loads and a small number of waiting places.

In order to increase the operating efficiency of networks with KSO in the high-load range, in the existing networks the method of limiting the nonproductive loading as a result of waiting calls is used. This is achieved by limiting the time spent by the call in the queue -- τ_{max}. In [2] some results from simulating such algorithms are presented for different values of τ_{max}. From the point of view of the problem stated in this paper it must be noted that the application of this method has not led to improvement of the quality indices of the network.

Inasmuch as the investigated type of overload occurs not only as a result of "long-waiting" calls, but also as a result of the entire mass of calls waiting in the tandem junctions, it is expedient to use a method that is simpler to implement to restrict the nonproductive load which consists in preventing waiting for tandem calls. In the presence of categorizing of calls it is possible to forbid calls of only the lowest categories from waiting. The expediency of using the given method is illustrated by the method of statistical simulation in [2].

However, the application of the indicated method permits only partial solution of the overload problem in a network with KSO and an increase in its carrying capacity. As is clear from what has been investigated, when using combined servicing systems in the KK networks, the following contradictions arise. Under conditions of low intensities of incoming flows, the use of bypasses is very efficient, and increasing the number of waiting places reduces the network losses for small values of the average waiting time; for high intensities of the incoming flows overloads and general network losses occur, and the average waiting time increases sharply. In addition to the above-noted cause of overloads, efficient use of bypasses has significance also in a network with rejects.

In (3), the problem of finding the dynamic control algorithm which is efficient in the entire loading range, that is, realizes expedient distribution of the call traffic with low load intensities and limits the load for high load intensities, is solved. The "efficient" algorithm was obtained on the basis of generalizing the "efficient control" principles to the case of networks with KSO. The cost of the k-th path of the j-th flow $D(\ell_j^k)$ for a system with KSO can be selected as follows:

$$D(\ell_j^k) = d_1(i) + \sum_{\substack{\xi \in \ell_j^k \\ \xi \neq 1}} \bar{d}_\xi \,, \tag{1}$$

where $d_1(i)$ -- the cost of the system $M/M/n_1/m_1$ in the state i -- is estimated as the forecasted increase in number of lost calls as a result of busy time in the branch γ_1 of the channel or the waiting place; $\bar{d}_\xi$ is the mathematical expectation of the cost of the branch γ_ξ, $\xi \in \ell_j^k$, $\xi \neq 1$.

It must be noted that the cost of the path includes the instantaneous cost of the first branch of the path and mathematical expectation of the costs of the subsequent ones, for the expression was obtained under the assumption that the states of the successive branches of the path, in addition to the first, is unknown at the time of arrival of a call for which prediction takes place. This is significant for a branch with waiting, for the call can wait in the process of being set up for a random time in each tandem UK. In the case of routing the call, a bypass path of minimum cost is selected under the condition that all the channels and all the waiting places are busy on the direct path. If $D(\ell_j^k) \geq 1$, the call is rejected. As statistical simulation has demonstrated, as a result of the application of an efficient algorithm on a symmetric network for m = 2 it was possible to cut the losses in the entire load range approximately in half by comparison with the static distribution. This reduction in losses is felt most significantly in the high-load range. This made it possible to increase the carrying capacity of the network with a specific load of 0.8 Erlang by 1 percent; with a load of 0.9 Erlang by 6 percent and with a load of 1.0 Erlang, by 10 percent.

It is possible to simplify the efficient algorithm. From the equality

$$D(\ell_j^k) = E_{n_1,m_1}(Y_1)/E_i(Y_1) + \sum_{\substack{\xi \in \ell_j^k \\ \xi \neq 1}} \bar{d}_\xi = 1, \qquad (2)$$

where $Y_1 = \lambda/\mu$ is the load intensity coming to the branch γ_1 of the path ℓ_j^k, under the condition that the calls are distributed only with respect to the direct routings, $E_{n_1,m_1}(Y_1)$ is the probability that the system $M/M/n_1/m_1$ will be busy; $E_i(Y_1)$ is the probability that the group of i channels will be busy for $i \leq n_1$, for $i > n_1$ the probability that the system $M/M/n_1/i - n_1$ will be busy.

It is possible to find the i which satisfies condition (2). Then if on arrival of a call on the bypass path ℓ_j^k on the branch γ_1 the number of servicing devices is less than i, the call is serviced. Otherwise a reject is received. The results of simulating such a simplified algorithm demonstrated that the servicing quality will be very close to the results obtained when using the theoretical algorithm.

A study was made of the dependence of the quality indices for the above-indicated network on the incoming load with an increase in number of waiting places to 10 when using the efficient algorithm. Inasmuch as its application offers the possibility of obtaining losses always less than when using only the forward connections, with an increase in number of waiting places and simultaneous use of the efficient control it is possible to obtain an increase in network carrying capacity both for low and high loads. The possibility of using an efficient algorithm combined with using the KSO to increase the carrying capacity of the KK network were limited only by the requirements on the maximum admissible average waiting time by a call for beginning of servicing.

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REDISTRIBUTION OF PROBLEMS IN A MICROPROCESSOR CONTROL NETWORK WITH FAILURES

[Article by Ye. N. Turuta, Moscow , pp 145-149]

[Text] One of the prospective approaches to solving the problems of increasing the reliability of distributed multimicroprocessor control systems is the approach based on ensuring the property of gradual degradation of the system in the case of failures of its processor modules. By this property we mean the capacity of the system to reconfigure and redistribute the problems of the failed modules among the working modules.

This redistribution offers the possibility of keeping the system operating as a whole in the case of failures of positive modules with worse, but admissible values of some of the operating indices of the system such as output capacity, functioning efficiency, and so on.

A study is made of a distributed multimicroprocessor technological process control system implemented in the form of a microprocessor control network (Figure 1). The functional processor modules (PM) solving the technological process control problems realize control information and data exchange among each other and with the objects of control (OU) by means of the communication network, the junctions of which are the processor switching modules (KPM).

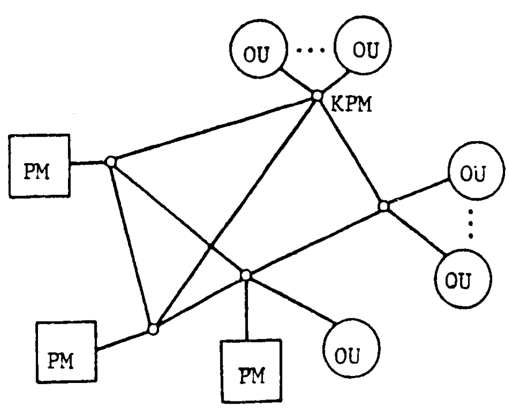

Figure 1.

Let us assume that PM failures are possible during operation of the system. The failures of the communication network channels, KPM and objects of control are not considered.

The state of the system at an arbitrary point in time is determined by the set of states of its PM: $\kappa_p = \sigma_1, \ldots, \sigma_n$, where $\sigma_i \in \{0, 1\}$ and $\sigma_i = 0$ if the PM_i is in the fit state; $\sigma_i = 1$ if PM_i is in the failed state.

[Two lines missing from the original text] ... processes solved in the system in the state κ_ν; $\Omega_i^\nu = \{\sigma_1^{i\nu}, \ldots, \sigma_\mu^{i\nu}, \ldots, \sigma_{m_{i\nu}}^{i\nu}\}$, $i = 1, \ldots, g_\nu$ is the subset of problems solved by the working PM_i in the state κ_ν, where $\sigma_\mu^{i\nu} \in \Omega^\nu$, $g_\nu - n = k_\nu$ is the number of working PM in the state κ_ν, k_n is the number of failed PM in this state, $\Phi_\nu = \{\phi_1^\nu, \ldots, \phi_{L_\nu}^\nu\}$ is the set of all technological processes (TP) taking place in the objects of control in the state κ_ν, $\nu = 1, \ldots, 2^n - 1$ inasmuch as fit PM are absent in the state $\kappa^1 = 11 \ldots 1$.

Organization of the redistribution of the problems of the failed PM requires execution of the following basic steps.

1. Finding the sets $\Omega^0 = \{\sigma_1, \ldots, \sigma_N\}$, $\Omega_i^0 = \{\sigma_1^i, \ldots, \sigma_\mu^i, \ldots, \sigma_{m_i}^i\}$, $\Phi^0 = \{\phi_1, \ldots, \phi_L\}$ corresponding to the state $\kappa^0 = 00 \ldots 0$.

2. Determination of the correspondence between the set of all control problems and the set of all TP for the state κ^0. This correspondence can be given by the matrix $\|a_{(\mu_i)k}\|$ having $R = \sum_{i=1}^{n} m_i$ rows and L columns where $a_{(\mu_i)k} = 1$ if for performance of the TP ϕ_k it is necessary to solve the problem σ_μ^i and $a_{(\mu_i)k} = 0$ otherwise.

3. The choice of indices on which defined requirements are imposed with organization of redistribution of the problems.

Let us consider the case where such indices are the system efficiency, the output capacity of the PM and the expenditures required for realizing the possibility of redistribution of the problems.

As the efficiency index of the system in the state κ_ν let us take the value $E_\nu = \sum_{\phi_k \in \Phi_\nu} \rho_k$, where ρ_k is the weight of the TP ϕ_k given by the requester.

We shall estimate the duration of the PM_i by the mean request servicing time for the performance of any mission $\sigma_\mu^i \in \Omega_i^\nu$ in the PM_i in the state κ_ν.

Let us assume that the average time T_i^0 for servicing a request in the PM_i in the state κ^0 and the matrix $\|\Delta T_{ji}\|$, $j = 1, \ldots, N$ and $i = 1, \ldots, n$ is given, where ΔT_{ji} is the increment of the average request servicing time in the PM_i on transfer of the problem σ_j to it. The required expenditures can be given by the matrix $\|C_{ji}\|$, $j = 1, \ldots, N$ and $i = 1, \ldots, n$, where C_{ji} are the expenditures which are required to ensure the possibility of transfer of the problem σ_j to the module PM_i (where $\sigma_j \in \Omega_\ell^0$) on failure of PM_ℓ. Let us assume that the value of C_{ji} does not depend on in which PM_ℓ the problem σ_j is realized in the state κ^0. On transfer of the problem $\sigma_j^\ell \in \Omega_\ell^0$ of the failed PM_l to the working PM_i two cases are possible: [four lines missing from the original text] ... that the PM_i will now carry out the problem beginning not only with its own requirements, but also the requirements of PM_ℓ.

In case (2), transfer of problem σ_μ^ℓ to modulus PM_i involves defined expenditures connected with the necessity for introduction of additional reserves into this modulus. Therefore we assume that $C_{ji} = 0$ if $\sigma_j \in \Omega_i^0$ and $C_{ji} \neq 0$ if $\sigma_j \notin \Omega_i^0$. In the special case the magnitude of the expenditures C_{ji} cannot depend on the PM_i to which the problem is transferred (for example, if all PM in the system are the same). Then $C_{ji} = 0$ if $\sigma_j \in \Omega_i^0$ and $C_{ji} = C_j$ if $\sigma_j \notin \Omega_i^0$ for any $i = 1, \ldots, q_\nu$.

4. Finding the set $S = \{G_\nu\}$ of distributions of the problems each of which corresponds to the state κ_ν from the given subset $\Gamma = \{\kappa_\nu\}$ of states of the system and satisfies the requirements imposed on the selected indices.

Two approaches to organizing the redistribution of the problems in the system in the case of PM failures are possible. The first approach assumes that the optimal distributions G_ν for all states $\kappa_\nu \in \Gamma$ (where Γ is the given subset of states) are found when designing the system and each PM is provided with the necessary reserves to carry out the problems required both in the state κ^0 and in each of the states $\kappa_\nu \in \Gamma$ in accordance with the distributions G_ν found when building the system.

The second approach is based on the fact that the problem of finding the optimal distribution G_ν is solved each time (using a special decision module) for transition of the system to the state κ_ν, and the result of its solution can depend on the preceding states of the system.

In the general case obtaining optimal distribution can require redistribution of the problems of both the failed and operating PM.

Let us consider the first approach to organizing the redistribution of problems, assuming that on transition of the system to the state κ_ν, the problems of only the failed PM are redistributed.

For each state $\kappa_\nu \in \Gamma$ we find the distribution G_ν satisfying the conditions:

$$E_\nu = \max,$$

$$C_\nu \leq C_{\nu\ ad},$$

$$T_i \leq T_{i\ ad},\ i = 1, \ldots, g_\nu,$$

where C_ν are the expenditures required to ensure the possibility of transferring the problems of the failed PM to the operating PM in the state κ_ν; T_i^ν is the average request servicing time in the PM_i in the state κ_ν; $C_{\nu\ ad}$, $T_{i\ ad}^\nu$ are the admissible values of the indicated indices for the state κ_ν.

The problem σ_j carried out for the initial distribution in the PM_i will be denoted by σ_{ji}. Let the initial distribution G_0 be given by the matrix $\|\beta_{ji}\|$, $j = 1, \ldots, N$ and $i = 1, \ldots, n$, where β_{ji} is the weight of the problem σ_j carried out for the initial distribution in the PM_i defined as

$$\beta_{ji} = \sum_{\varphi^s_{ji} \in \Phi_{ji}} \rho^s_{ji},$$

where $\Phi_{ji} = \{\phi^s_{ji}\}$ is the set of TP for performance of which it is necessary to satisfy the problem σ_{ji}, ρ^s_{ji} is the weight of the TP ϕ^s_{ji}.

Let $D^*_\nu = \{\Pi M^\nu_\ell\}$, $D_\nu = \{\Pi M^\nu_i\}$ be the set of failed and operating PM respectively for the state κ_ν; A^*_ν be the set of all problems which in the state κ^0 are carried out in the modules $\Pi M^\nu_\ell \in D^*_\nu$.

The component of the system efficiency in the state κ_ν determined by the performance by the operating PM in this state of the problems $\sigma_{j\ell} \in A^*_\nu$ can be expressed as

$$E^*_\nu = \sum_{i=1}^{q_\nu} \sum_{\alpha_{j\ell} \in A^*_\nu} d^i_{j\ell} \beta_{j\ell},$$

where $d^i_{j\ell} \in \{0, 1\}$ and $d^i_{j\ell} = 1$ if the problem $\sigma_{j\ell}$ is transferred to the PM_i, $d^i_{j\ell} = 0$ otherwise.

It is possible to assume that finding the optimal distribution G_ν for fixed state κ_ν reduces to solving a problem of integral linear programming consisting in finding the set of values of the variables $d^i_{j\ell}$ maximizing the value of the function E^*_ν under the restrictions:

$$C_\nu = \sum_{i=1}^{q_\nu} \sum_{\alpha_{j\ell} \in A^*_\nu} d^i_{j\ell} C_{(j\ell)i} \leqslant C_{\nu\, ad},$$

$$T^\nu_i = T^0_i + \sum_{\alpha_{j\ell} \in A^*_\nu} d^i_{j\ell} \Delta T_{(j\ell)i} \leqslant T^\nu_{i\, ad}, \quad i = 1, \ldots, q_\nu,$$

where $C(j\ell)_i$ are the expenditures required for realizing the transfer of the problem $\sigma_{j\ell}$ to the modulus PM_i (given by the matrix $\|C_{ji}\|$); $\Delta T(j\ell)_i$ is the increment of the average servicing time of a request in the operating PM on transfer of the problem $\sigma_{j\ell}$ to it (given by the matrix $\|\Delta T_{ji}\|$).

This problem can be solved by any of the known TsLP methods [2].

5. Finding the problems which each PM must solve in the state κ^0 and in all states $\kappa_\nu \in \Gamma$ and calculation of the total expenditures required to realize the obtained set $S = \{G_\nu\}$ of distributions, the average request servicing time in each PM and the values of the efficiency index of the system for each of the states of the given set $\Gamma = \{\kappa_\nu\}$.

The optimal distribution of the problems found for each state $\kappa_\nu \in \Gamma(\nu = 1, \ldots, r)$ is described by the matrix $\|A^\nu_{ji}\|$, where $j = 1, \ldots, N$ and $i = 1, \ldots, n$, A^ν_{ji} is a subset of problems defined below.

For the columns corresponding to the operating PM_i in the state κ_ν $A^\nu_{ji} = \sigma_{ji} \cup \{\sigma_{j\ell}\}^{\nu i}$, where σ_{ji} is the problem executed in the PM_i in the state κ^0; $\{\sigma_{j\ell}\}^{\nu i}$ is the set of problems $\sigma_{j\ell}$ transferred to the PM_i in the condition κ_ν from failed PM_ℓ, where they are carried out in the state κ^0.

For columns corresponding to the failed PM_ℓ $A^\nu_{ji} = \Phi$. Let us construct the resultant matrix $\|A_{ji}\|$, where $A_{ji} = \bigcup_{\nu=1} A^\nu_{ji}$. The matrix $\|A_{ji}\|$ defines the set of problems for each PM_i, $i = 1, \ldots, n$, which must be carried out in it in the case of failures of the given subsets of PM in order that in each of the corresponding states of the systems the maximum value of its efficiency will be reached for given restrictions on the output capacity of the PM and additional expenditures. The above-indicated system indices can be calculated by this matrix.

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MAN-MACHINE METHOD OF OBTAINING AN ALGORITHM FOR THE OPERATION OF A CONTROL UNIT

[Article by L. K. Yan'shina, Moscow , pp 149-152]

[Text] The initial step in the synthesis of microprogram control units is the step of obtaining the operating algorithm (AF) in the formal synthesis language. The difficulties of this step consist in the fact that the requester, as a rule, is not familiar with the formal synthesis languages and cannot competently write the algorithm for the behavior of his device in them. In a number of existing automated design systems (ASP) of digital control units these difficulties are overcome by creating specialized input languages oriented toward a specific class of users and similar with respect to their dictionary composition and grammatical rules to the terminology adopted in the given field of engineering. However, even in this case it remains necessary for the user to study these highly complex and formalized languages. In the automated design system for microprogram automata (ASPUMA), another procedure is proposed for obtaining an algorithm for the functioning of the control unit not requiring any formalized languages of the user. It is only assumed that he has a meaningful description of the AF of his device.

Obtaining the AF of the control unit in the ASPUMA system is an interactive process, the active side in which is the ASP (computer). However, it is highly complicated to realize adequate translation of the meaningful description of the AF to the language of logical flowcharts for algorithms used as the design language in ASPUMA directly. Therefore first the AF is written in an intermediate language (the conversion formula language [1] or sequence language), and then this notation is translated to the language of the logical flowcharts of algorithms (LSA).

The ASP initiates problems of two types: the problem requiring a descriptive answer, the problem requiring an alternative answer ("yes-no" type). The answers to problems of the first type are given in natural Russian language without any requirements on syntax; the answers to problems of the second type are regulated. The dialogue is realized by a computer videoterminal, and it contains the following steps.

1. Obtaining the set of control inputs.

2. Explanation of the control input feed sequence.

3. Description of the set of logical conditions.

4. Discovery of the values of the logical conditions.

After some familiarization with the technique of performing the dialogue and the capabilities of the ASP, the requester sets to work on the first step of the dialogue procedure.

1. Obtaining the set of control inputs:

In this step it is assumed that the requester will describe all of the control inputs required for proper operation of his device. He must describe the first control input (operator), second, and so on. Each control input can be described by means of several Russian sentences and contain a brief characteristic of it. All of the described ASP operators are assigned notation with which the requester is familiar in the dialogue protocol. In the last steps of the dialogue procedure all of the described operators are identified only by introduced notation.

2. Explanation of the control input feed sequence:

It is proposed that the client consider all of the operators described by him successively. If the conversion formula language is used as the intermediate language, the requester must answer the question as to which operators can be performed after the given operator with observation of certain conditions. If the sequence language is used as the intermediate language, the requester must call out the operators which are performed before the given operator. Thus, for each operator either a set of follower operators will be obtained or a set of precursor operators. The choice of the intermediate language is left to the requester.

3. Description of the set of logical conditions:

In this step all conditions are discovered for each pair of operators between which transfer is possible, the checking of which is necessary when realizing the given transfer. The computer outputs the pair of operators A_i, A_j and explains to the requester what conditions must be checked so that the control input A_j can be sent after the control input A_i. Each logical condition (LU) must be described by several Russian sentences. The described LU obtains a description which is recorded in the dialogue protocol. After all possible transfers A_i, A_j are investigated and a set of LU is described for each transfer, the performance of the third step is completed.

4. Discovery of the values of the logical conditions:

In the intermediate languages used, only the two-valued logical conditions are considered which assume two values: true and false. In the given step of the dialogue the requester is given the LU for investigation with indication of the transfer $A_i \rightarrow A_j$ for which it is checked. He must answer the question as to whether the given transfer takes place on satisfaction of the specific LU (that is, for its true value) or nonsatisfaction of the LU (for a false value)? When all of the transfers from one operator to another are investigated and the values of all of the described LU are discovered, the writing of the operating algorithm in the intermediate language will be completed.

After the AF is obtained in the intermediate language, translation of it to the language of logical flow diagrams of algorithms is realized. Since different languages can be used as the intermediate languages, it is expedient to do the translation by syntactic methods. For the conversion formula and sequence languages, generative grammars have been constructed. Inasmuch as the number of rules of these grammars is small, the power of the terminal and nonterminal dictionaries is also small, and they are reduced by standard methods to precedence grammars, and the methods based on precedence ratios were used to construct the translator. The recognition element of this translator is very simple and universal for a class of input languages [2].

Each input language is represented by a table of generative rules and precedence matrix, and the output language is represented by a set of semantic programs.

In conclusion, it is necessary to note that the given procedure can be used to automate programming, that is, to obtain the block diagram or logical flow diagram of the algorithm of the complex program.

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